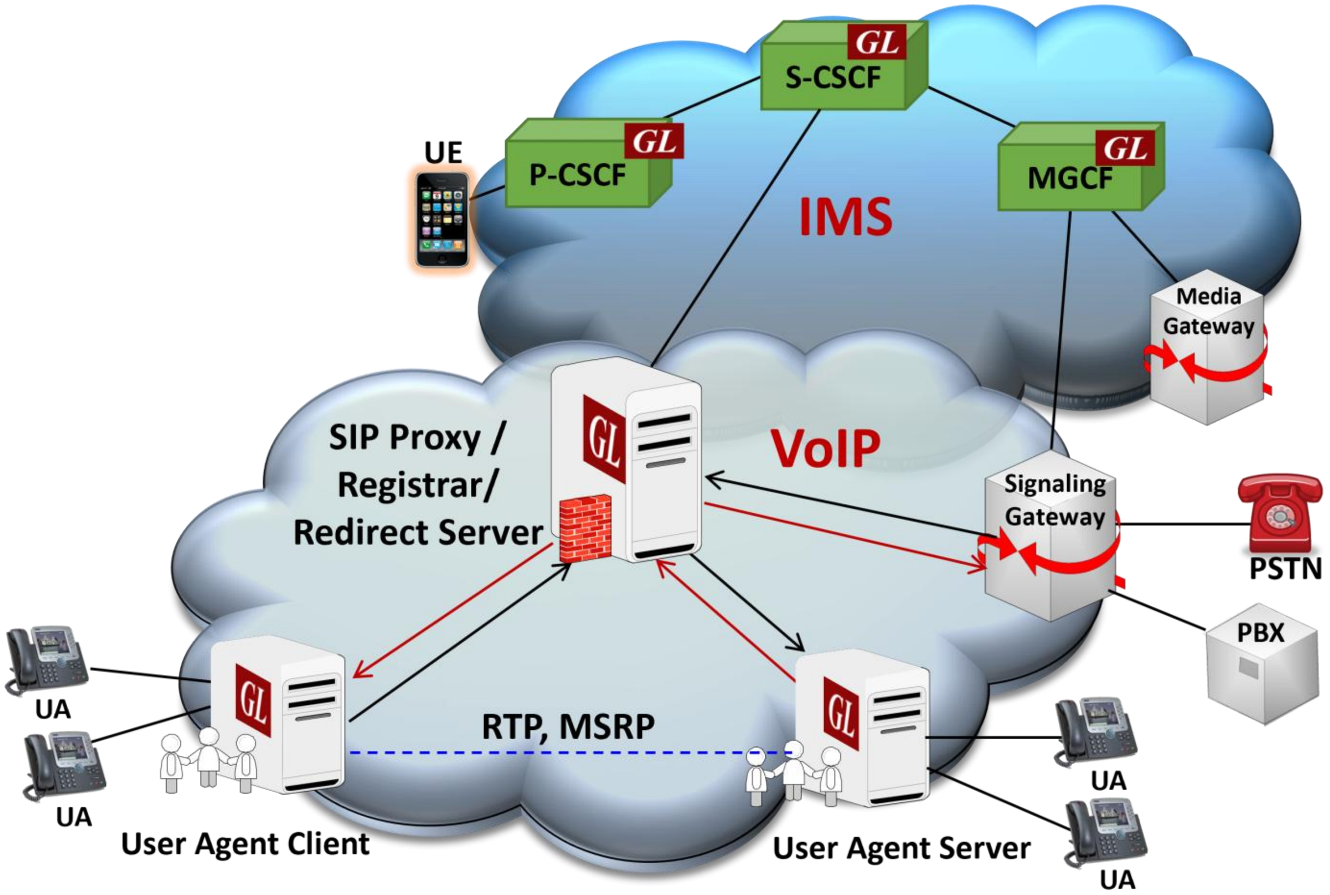
MAPS™ SIP
SIP + RTP + MSRP Simulation

8 April 2026

GL Communications Inc.
818 West Diamond Avenue - Third Floor, Gaithersburg, MD 20878
Phone: (301) 670-4784 Fax: (301) 670-9187 Email: info@gl.com
Website: https://www.gl.com

MAPS™ SIP



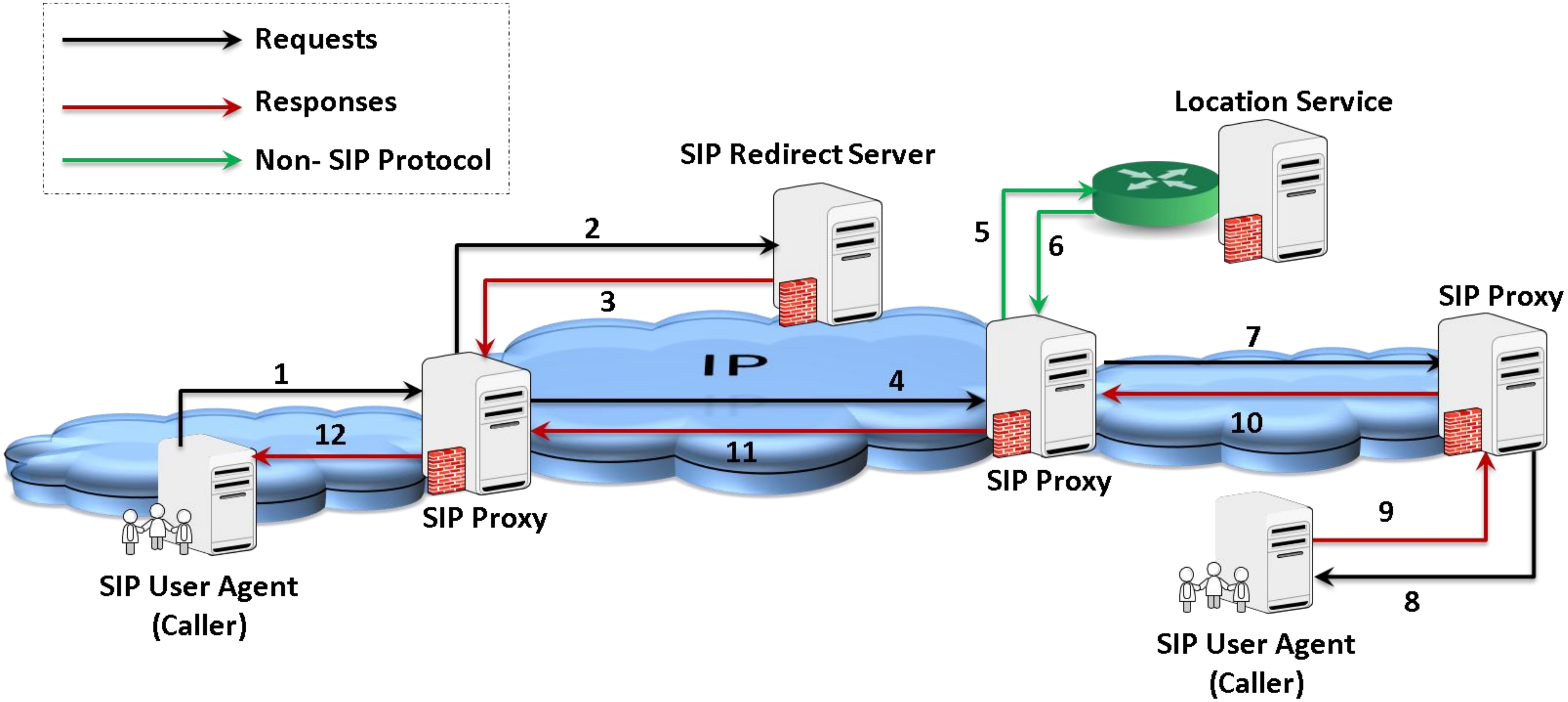
MAPS™ SIP with RTP Traffic Generation
(2500 simultaneous calls)

MAPS™ SIP Conformance

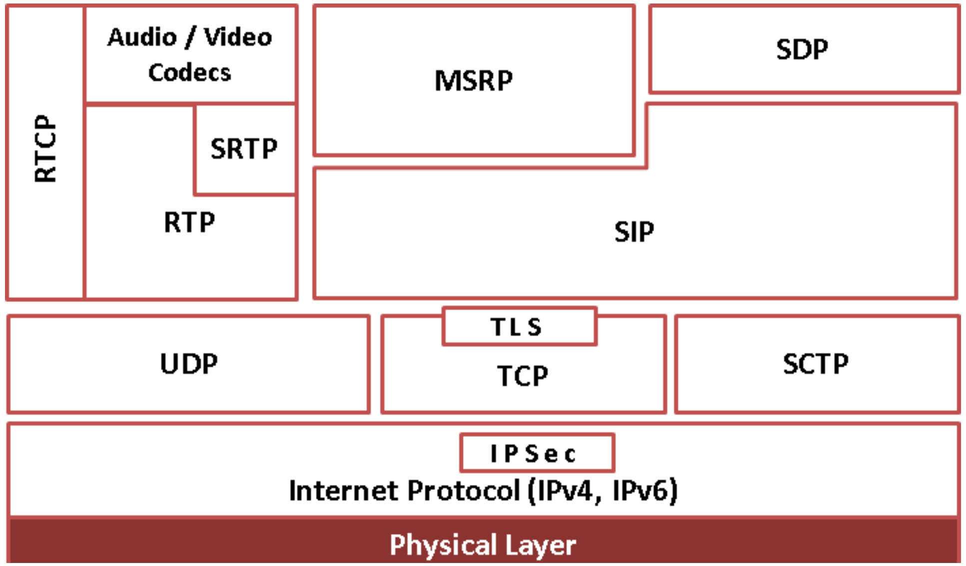
MAPS™ SIP (w/ 8 x 10 Gbps Ethernet Ports)
HD RTP Traffic Generator
160,000 Simultaneous Calls (with RTP Traffic)

GL Communications

SIP Architecture and Entities

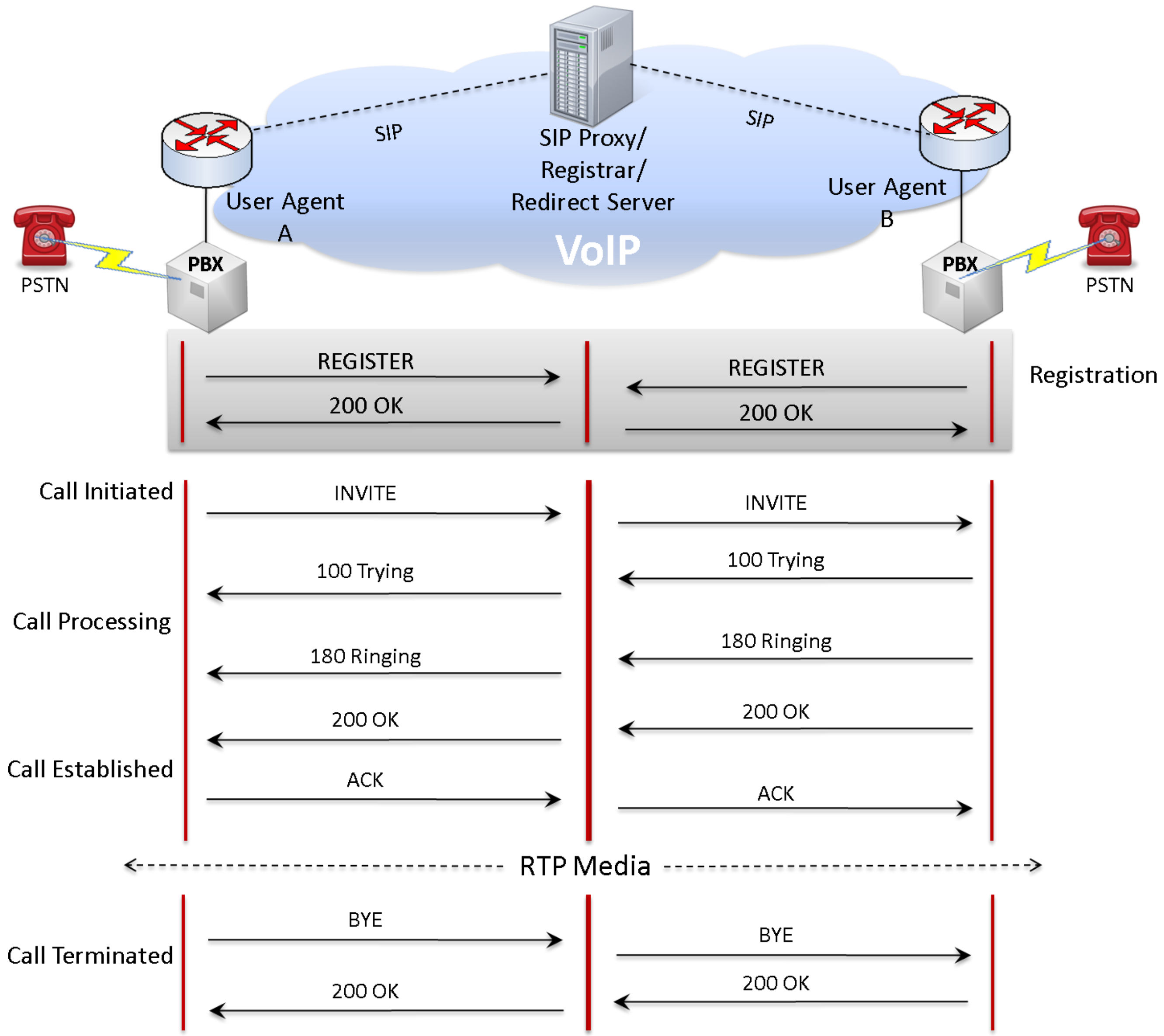


SIP Protocol Stack



Supported Protocols	Standard / Specification Used
SIP SIP Conformance	RFC 3261 ETSI TS 102-027-2 v4.1.1
SIP Extensions	RFC 3262 - Reliability of Provisional Responses in the Session Initiation Protocol (SIP) RFC 3311 - The Session Initiation Protocol (SIP) UPDATE Method RFC 3455 - Private Header (P-Header) Extensions to the Session Initiation Protocol (SIP) for the 3rd-Generation Partnership Project (3GPP) RFC 3515 - The Session Initiation Protocol (SIP) Refer Method RFC 3310 - HTTP/SIP Digest Authentication Using Authentication and Key Agreement (AKA) RFC 3263 - Session Initiation Protocol (SIP): Locating SIP Servers
Secure Real-time Transport Protocol (SRTP)	RFC 3711 - Secure Real-time Transport Protocol (SRTP) RFC 3551 - Standard 65, RTP Profile for Audio and Video Conferences with Minimal Control)
Message session Relay Protocol (MSRP)	RFC 4975 - Message Session Relay Protocol (MSRP)

Generic SIP Call Flow



GL Communications

MAPS™ SIP Variants

MAPS™ SIP Protocol Test Tool (Item # PKS120):

- RFC 3261 - Primary SIP standard
- RFC 3262 - PRACK
- RFC 3515 - REFER

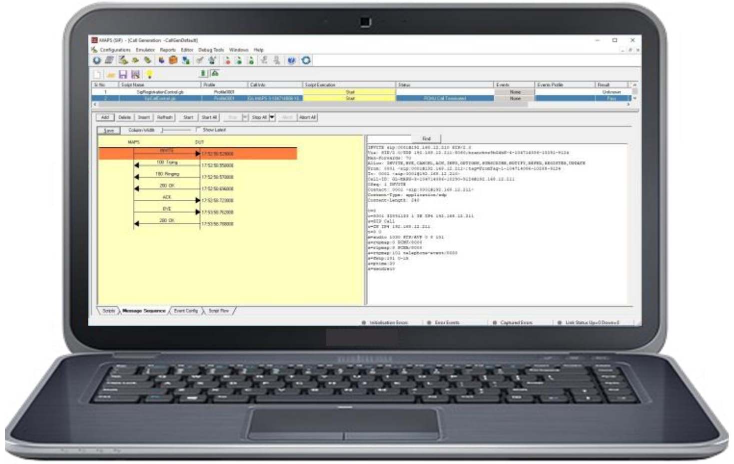
MAPS™ SIP Conformance Suite (Item # PKS121):

- ETSI TS 102-027-2 v4.1.1 (2006-07) - 300+ scripts designed to test SIP UAs for conformance to RFC 3261

MAPS™ SIP High Density (HD) (Item # PKS109):

- MAPS™ SIP HD emulates up to 160,000 simultaneous calls using 8 Gigabit Ethernet ports
- Available in portable or rack-mount form factors

MAPS™ SIP Software with Notebook PC



MAPS™ SIP HD

MAPS™ SIP Highlights

signaling	• Generates and processes SIP valid and invalid messages • Supports complete customization of SIP headers, call flow, and messages • Supports complete customization of scripts and parameters in the profiles • Each SIP message template facilitates customization of the protocol fields and access to the various protocol fields from the scripts • Supports IPv4 /IPv6 and transport over UDP and TCP, and TLS for secure transport • Handles Retransmissions of messages with specific interval • Scripted call generation and call reception • Supports 64-bit version to enhance signalling performance • Supports joining conference call, unattended call transfer, attended call transfer, call hold, auto call rejection, and silence packets generation • Ability to send "reliable provisional responses" and start early media actions • Ability to implement IP Spoofing for any network like Class C, Class B etc • Supports in dialog and out of dialog transactions for SUBSCRIBE, NOTIFY, OPTIONS, REFER and INFO SIP methods
Automation	• Automation, Remote access, and Schedulers to run tests 24/7 • Client-server model allows users to control all features of MAPS™ through APIs • Supported clients include Python and Java

MAPS™ SIP Highlights (Contd.)

Traffic	• Supports various RTP traffic (PKS102) such as, digits, voice file, tones, IVR, FAX, and Video in IP networks
	• Supports almost all industry standard voice codec types - G.722, G.729, G.726, GSM, AMR, EVRC, EVS, OPUS, SMV, iLBC, SPEEX, and more. *AMR and EVRC variants require additional licenses
	• Supports 64-bit RTP core to enhance performance - handles increased call rate of up to 3000 calls with high volume traffic.
	• Supports both G.711 Pass Through Fax Simulation (PKS200) and T.38 Fax Simulation over UDPTL (PKS211)
	• Generates RTP-based video traffic using pre-encoded video payloads
	• Study packet effects through impairment generation:
	  • Latency (Uniform distributed and Normal distributed)
	  • Packet loss (Periodic and Random)
	  • Packet effects (Duplicate and Out of order)
	• Supports Secure Real-time Transport Protocol (or SRTP) traffic initialized over TLS (Transport Layer Security) or SSL (OpenSSL)
	• User-defined voice quality statistics for received RTP Traffic can be calculated and updated periodically during run-time to a csv file
	• Supports simulation of SIP/MSRP User Agents end-points in an NG9-1-1 network and send and receive communications over IP networks. MSRP sessions supports simulation of IM Only Calls, Audio and IM Calls, and Video and IM Call types

GL Communications

SIP Call Types

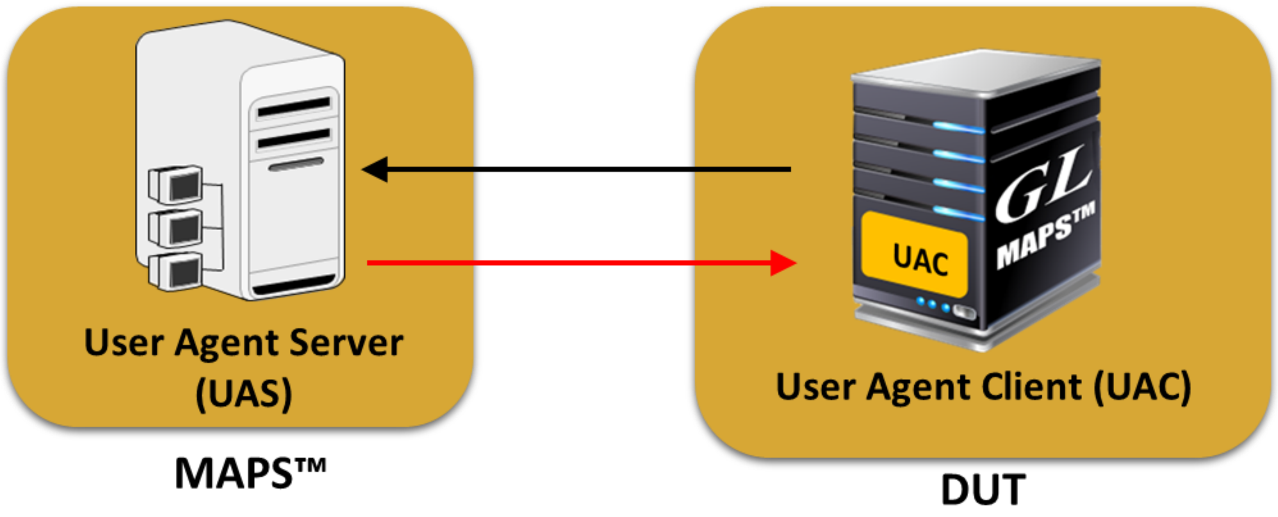
- Registration and Normal Call
- Call Redirection – Redirect the call to new location
- Call Transfer - Transfers the call using REFER Method
- Authentication – Challenging the incoming message for credential
- Early Media (PRACK support)
- Rejecting the call with Client Error (4XX), Server Error (5XX) and Global Error (6xx)

MAPS™ SIP Configured as UAS

Testing UAC

Scenario: MAPS™ acting as UAS and testing UAC

- MAPS™ acting as UAS receives messages from UAC (DUT)
- DUT (UAC) generates SIP messages

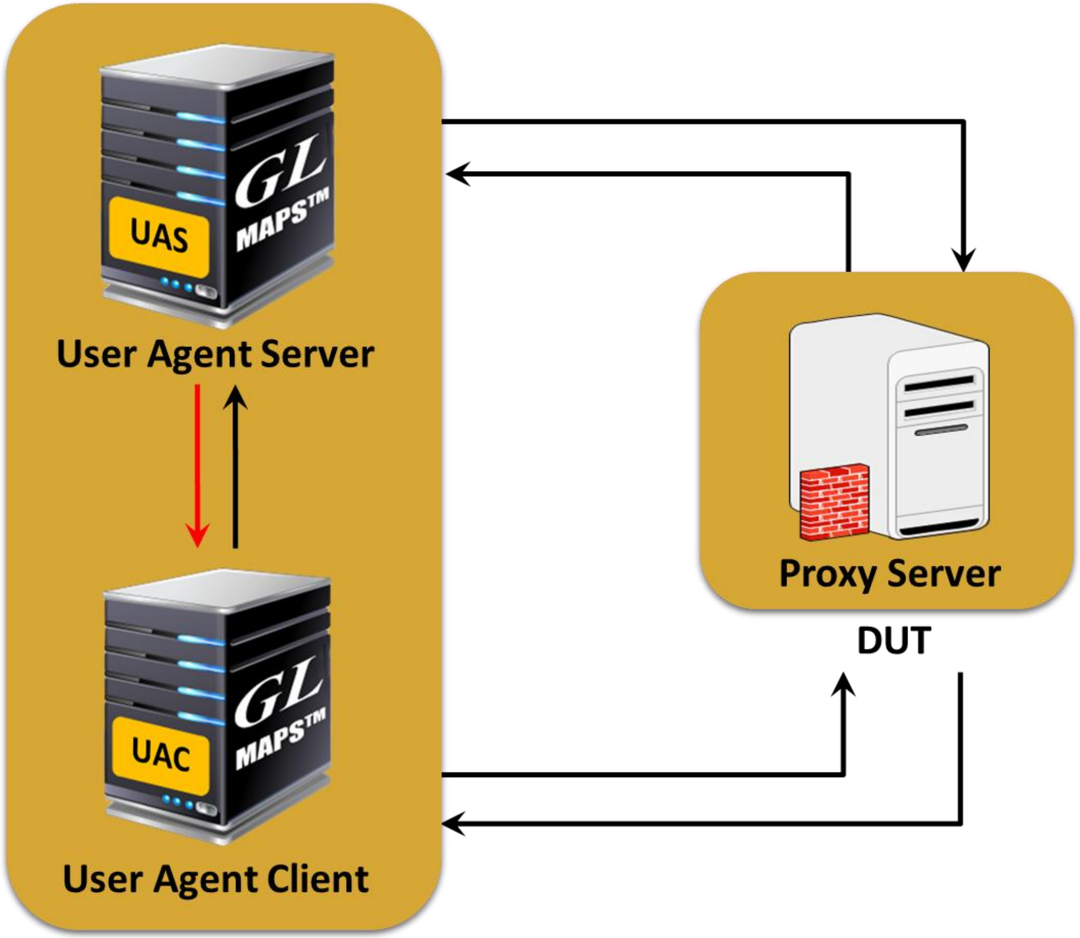


MAPS™ SIP Configured as UAC / UAS

Testing Proxy Server / B2B UA

Scenario: MAPS™ acting as UAS and UAC and testing Proxy.

- MAPS™ can be configured to act as UAC and UAS simultaneously so that entire Proxy testing can be automated

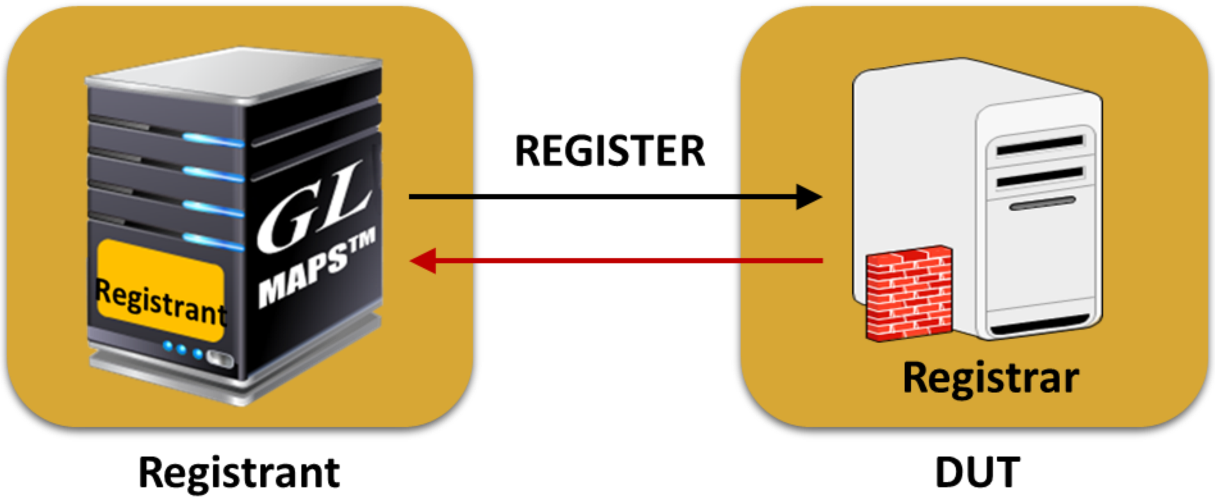


MAPS™ SIP Configured as Registrant

Testing Registrar

Scenario: MAPS™ acting as Registrant and testing Registrar.

- MAPS™ can be configured to act as Registrant and to generate registration request messages to automate the entire Registrar (DUT) testing

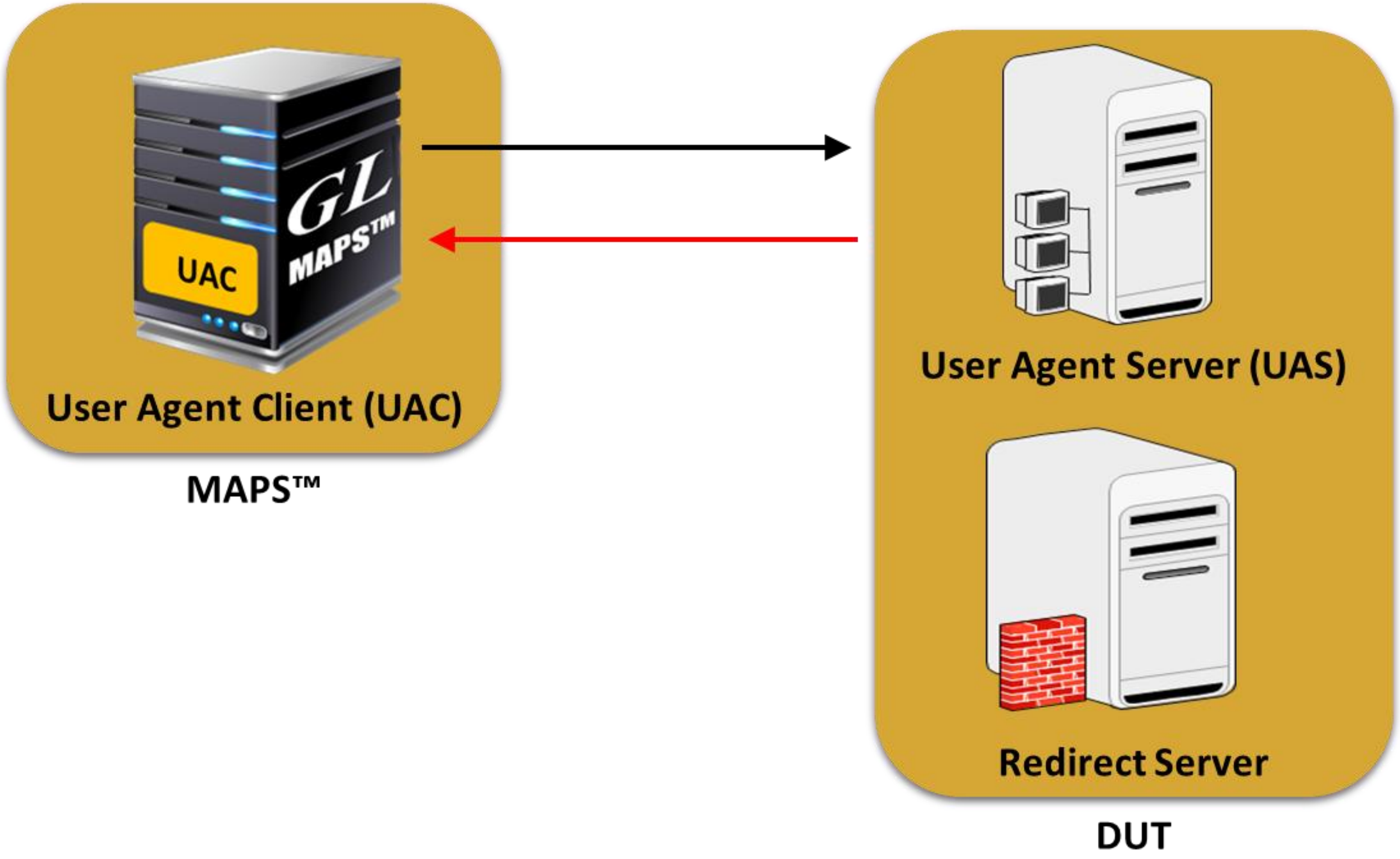


MAPS™ SIP Configured as UAC

Testing UAS & Redirect Server

Scenario: MAPS™ testing Redirect Server and / or UAS

- MAPS™ can be configured to act as UAC & generate SIP messages
- Tests Redirect Server and /or UAS; allows redirection of call scenarios to be automated

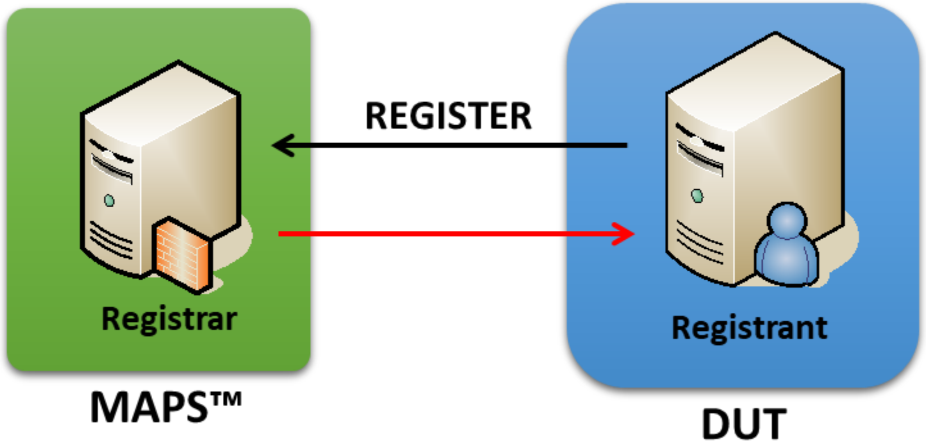


MAPS™ SIP Configured as Registrar

Testing Registrant

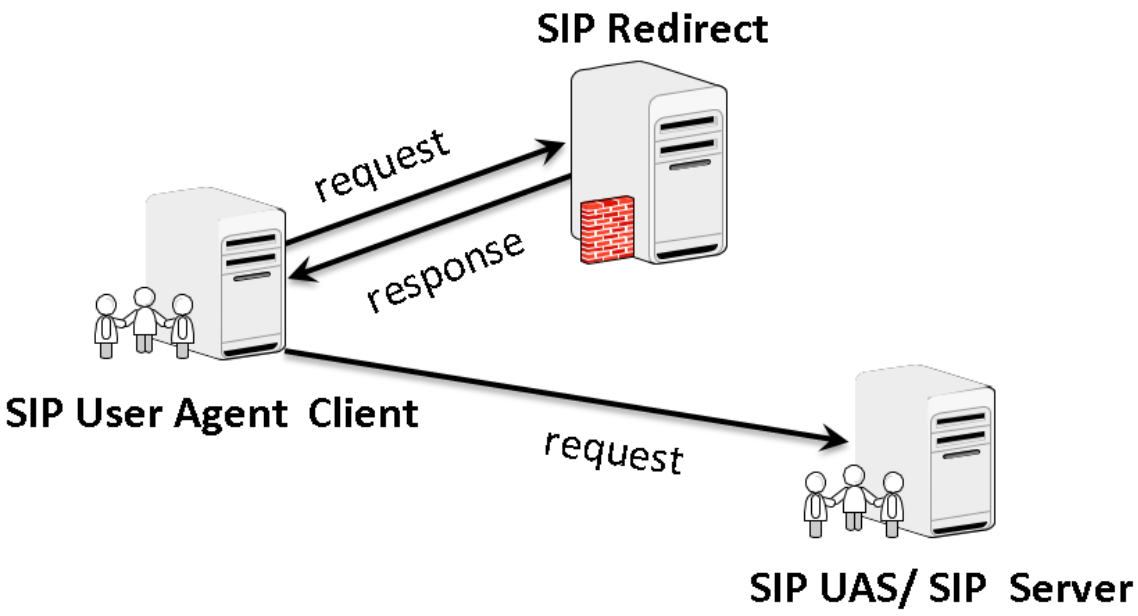
Scenario: MAPS™ acting as Registrar and testing Registrant

- MAPS™ acts as Registrar and processes received registration request messages from Registrant (DUT) while conforming Registrant
- DUT (Registrant) generates REGISTRATION SIP messages



SIP Redirect Server

- Returns the next address to originator instead of forwarding
- Originator retries with the new address



Call Generation (UAC)

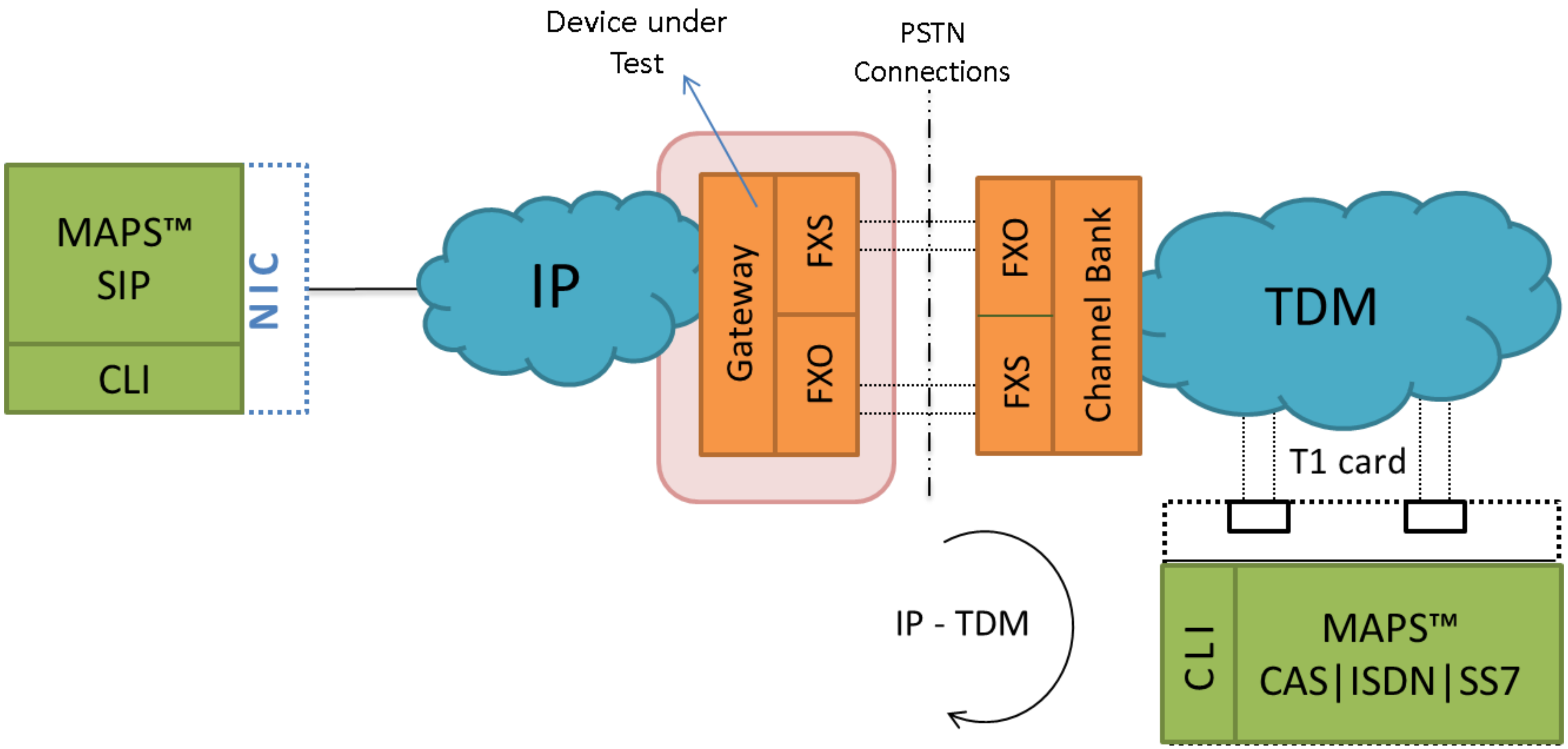
- Registrant – Registers with Registrar
- Call with Auto Traffic of RTP Action
- Traffic Impairments
- Simulates IVR (Interactive Voice Response) for RTP traffic
- Call through Proxy
- Sequential and Random Generation of Calls
- Simultaneous Generation of Calls
- Load Generation (Stress Testing)

Call Reception (UAS)

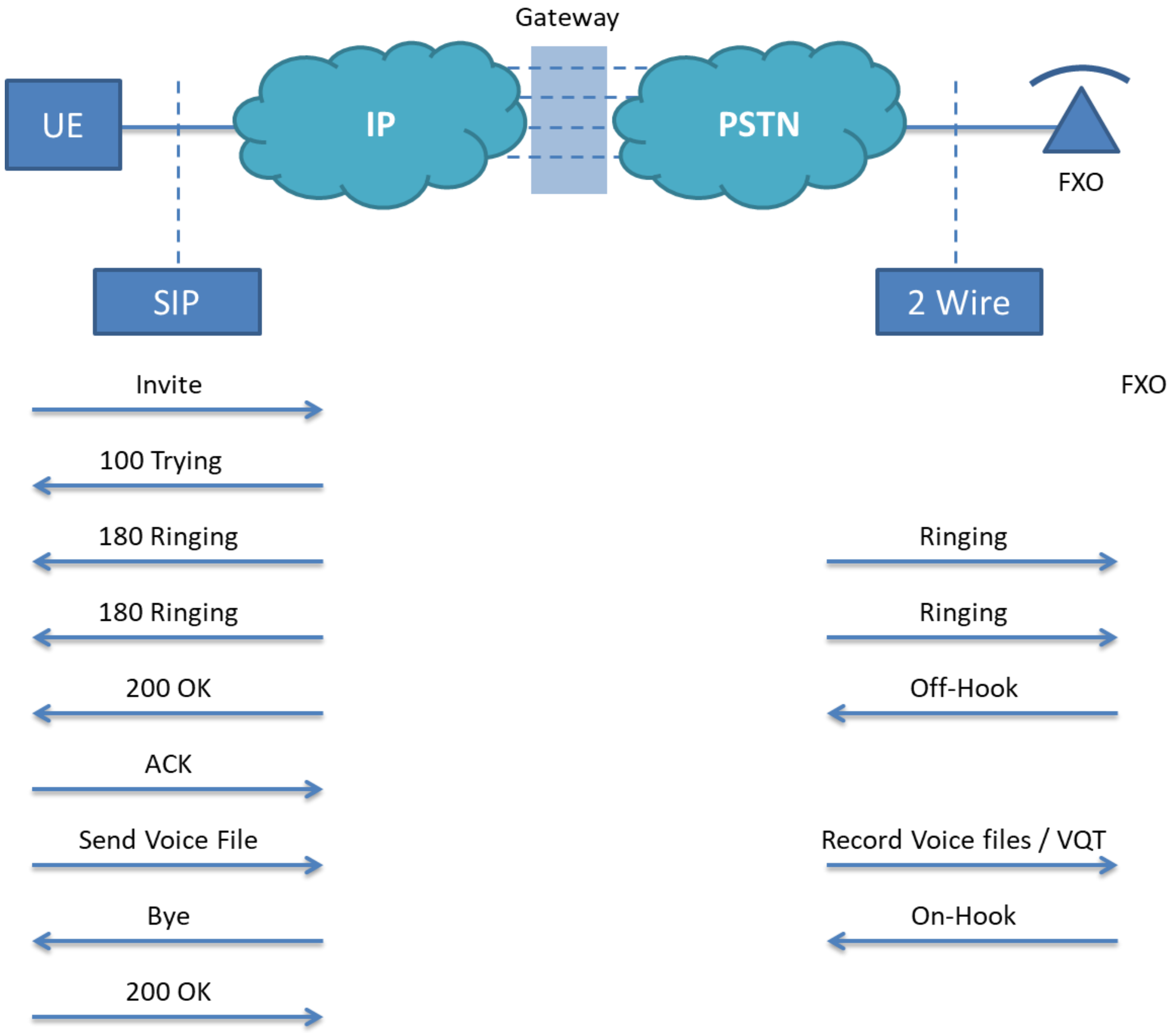
- Registrar – Accepts the registration from registrant
- Call Redirection – Redirect the call to new location
- Call Transfer - Transfers the call using REFER Method
- Authentication – Challenging the incoming message for credential
- Early Media (PRACK support)
- Rejecting the call with Client Error (4XX), Server Error (5XX), and Global Error (6xx)

End-to-End Gateway Testing

- Evaluates Gateway / ATA product features such as call connectivity, call signaling, traffic generation, voice quality testing, codec, and hundreds of other features

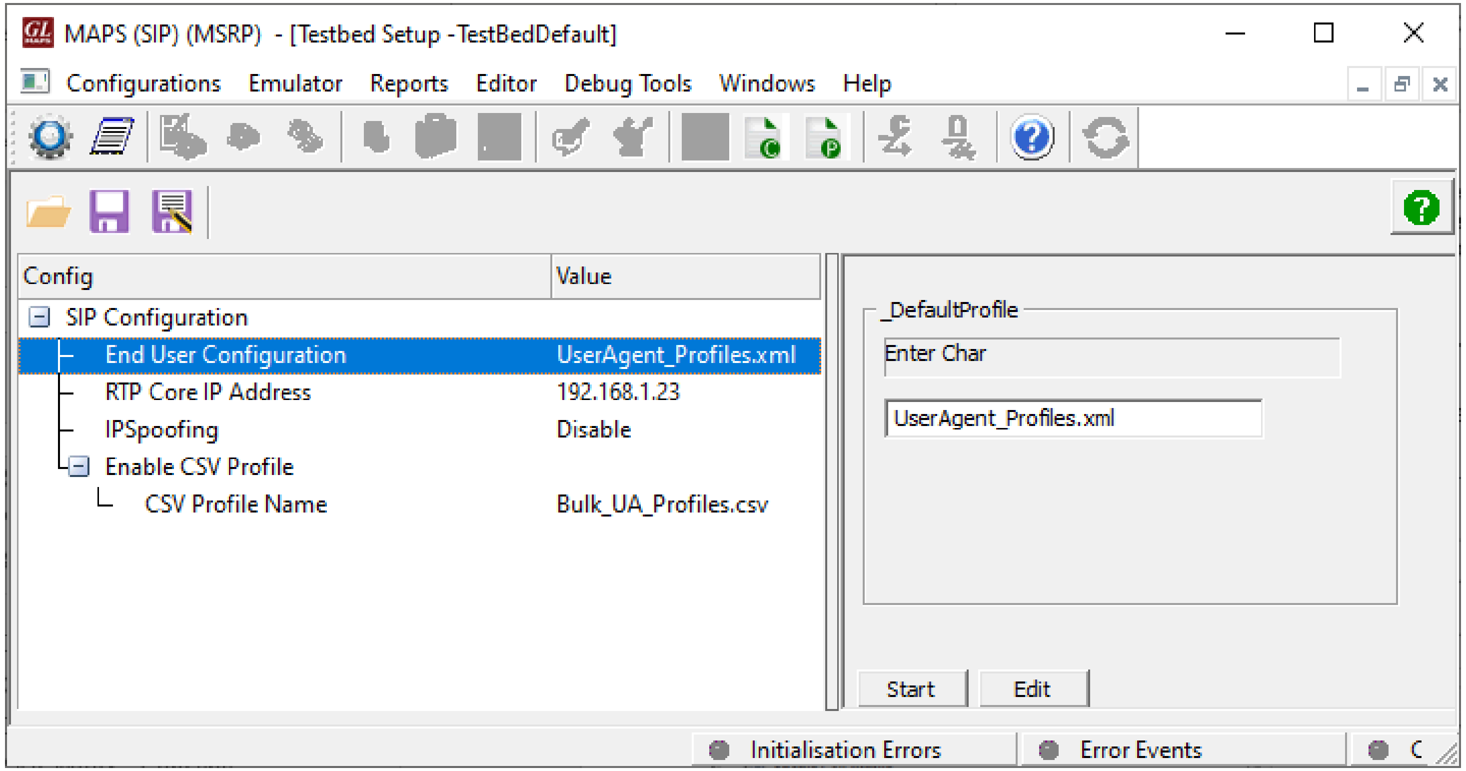


End-to-End Gateway Testing Call Scenario



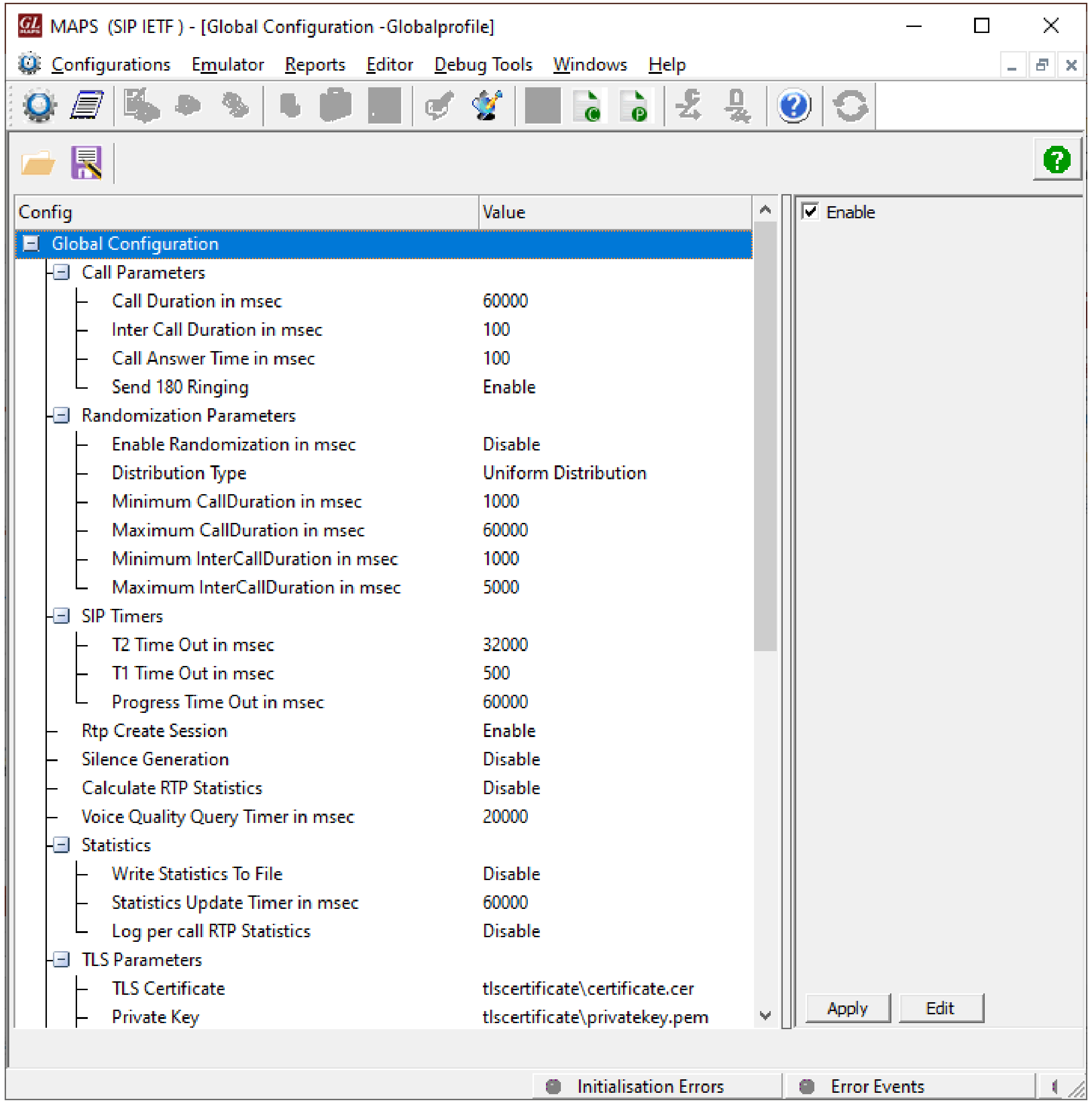
Test Bed Configuration

- **End User Configuration**: xml file containing one or more endpoint configurations
- **RTP Core IP Address**: IP Address of the system on which the RTP Core should be invoked
- **IP Spoofing**: permits user to assign one or more virtual IP addresses to NIC



Global Configuration

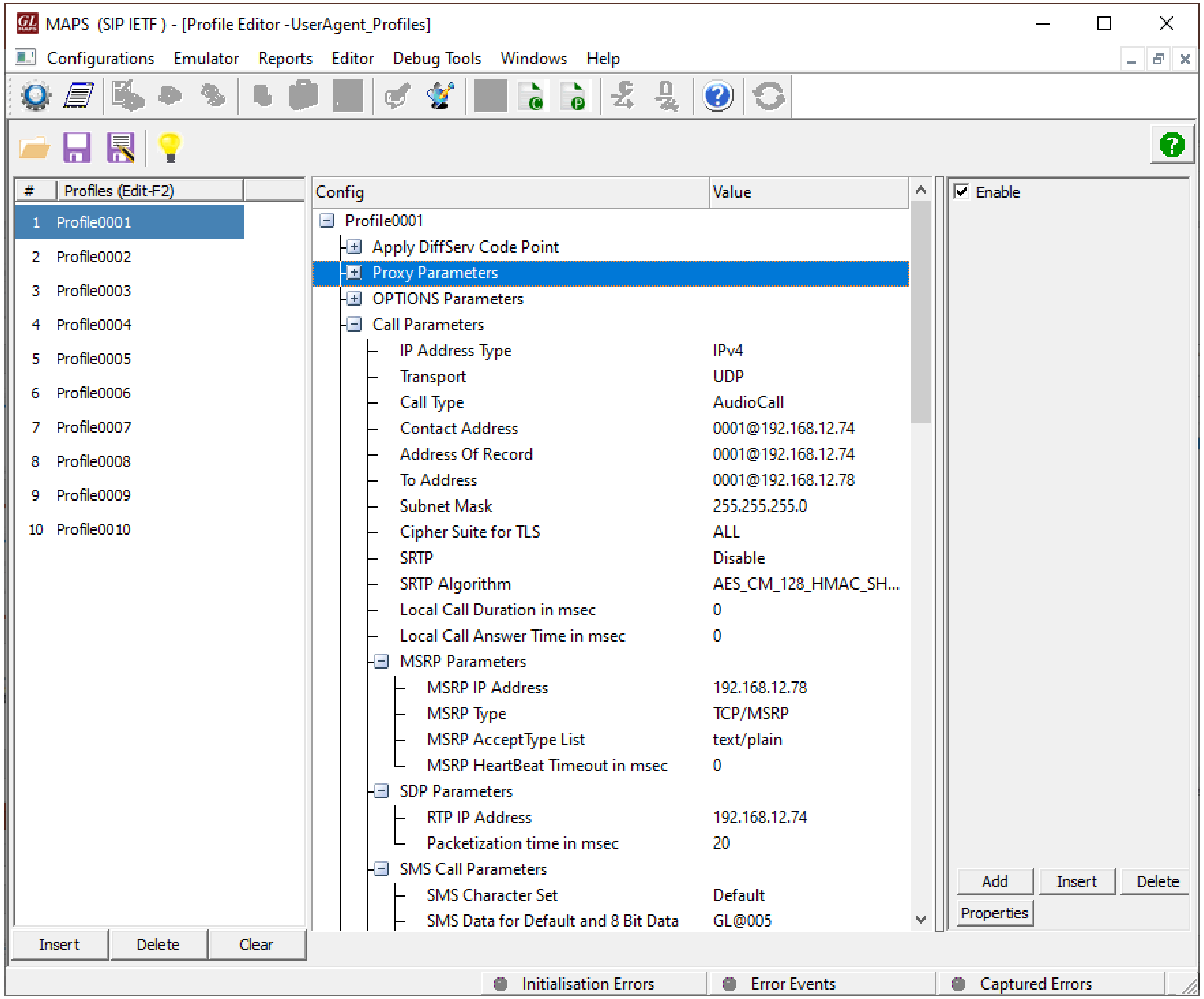
- A list of variables/values that are automatically declared and assigned at the start of any script execution
- A script may locally override the values assigned here
- A script may also ignore these variables entirely. For example, Call Duration is not a hard limit on the length of a call, it is just a variable the script may use



GL Communications

User Agents Configuration

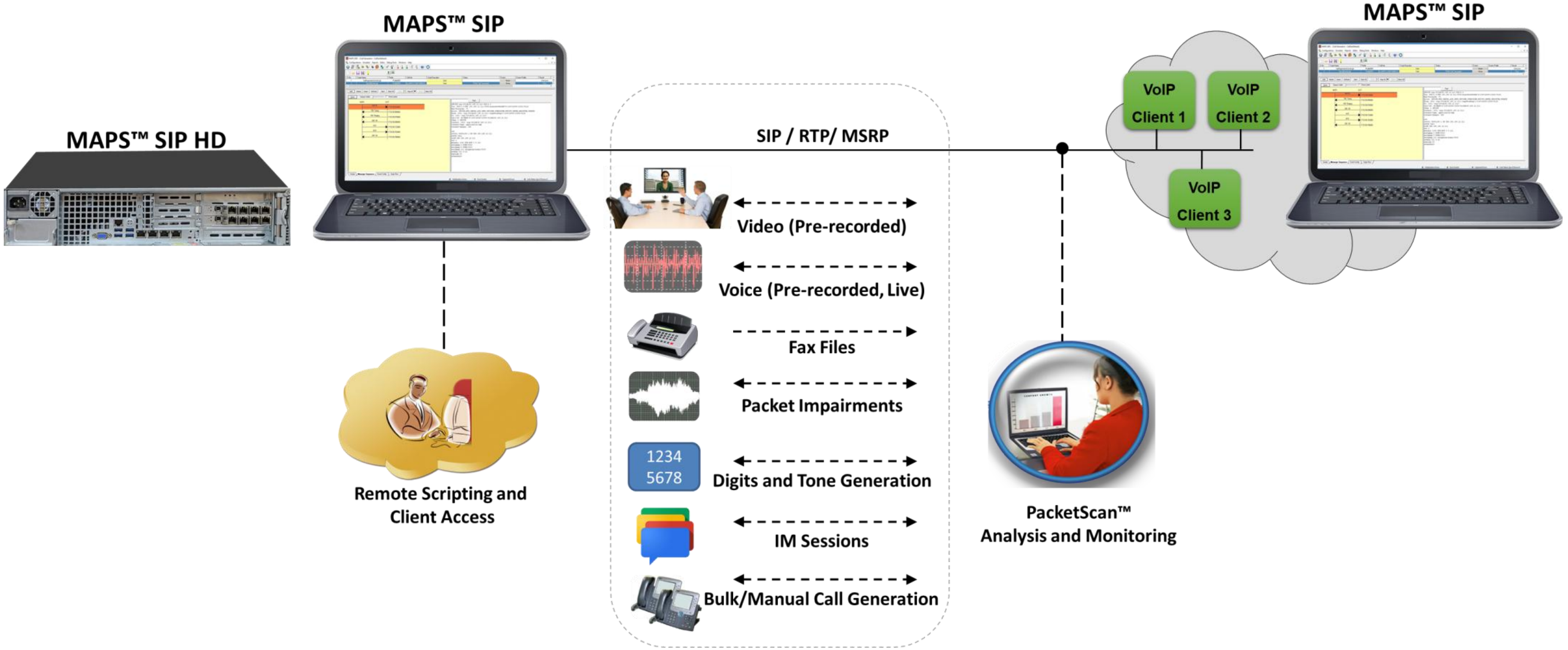
- Each Profile Group contains one or several sub-profiles
- Each sub-profile is a set of variables which together define a single SIP Endpoint
- Not every field in a profile is relevant to every script execution
- Profile Editor has a “Quick Config” tool to help users create multiple different sub-profiles



GL Communications

IP Traffic Simulation Capabilities and Performance

2000+ Simultaneous Calls (SIP + RTP Voice)
500 Simultaneous Calls (SIP + RTP Video)
500 Simultaneous Calls (SIP + IM MSRP)



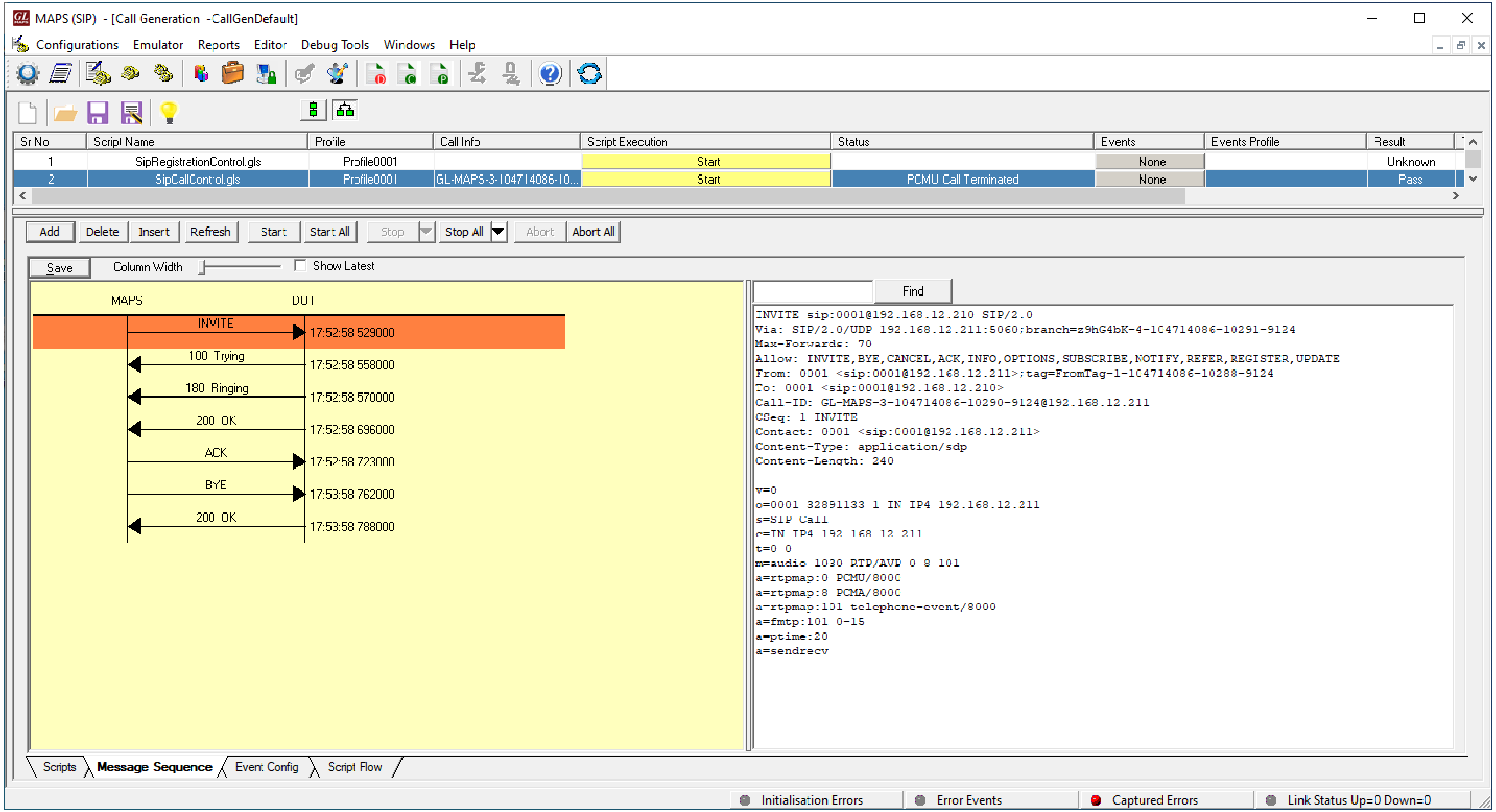
GL Communications

SIP Capabilities and Performance

Product Version	Max Simultaneous Calls			
	Only Signaling	Signaling + RTP Voice Traffic	Signaling + RTP VideoTraffic	Signaling + MSRP (IM) Traffic
MAPS™ SIP 64-bit (Core i7 with 12GB RAM)	30,000 Calls @ 250 CPS	2500 @ 250 CPS	500	500
MAPS™SIP HD 64-bit (Zeon Server with 16 Processors and 64GB RAM)	100,000 Calls @250 CPS	20000 @ 250 CPS	-	-

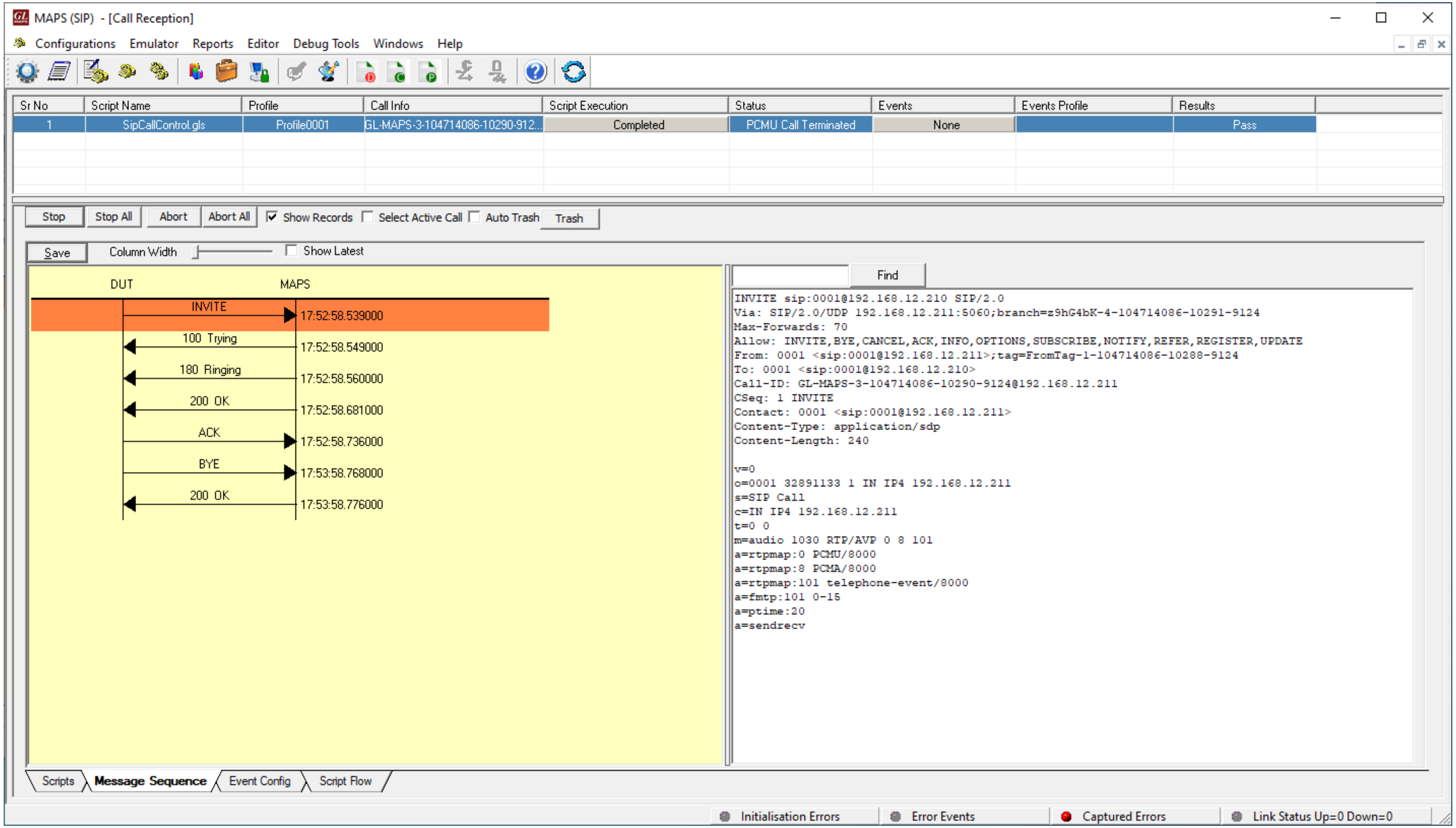
GL Communications

Call Generation with Voice Traffic



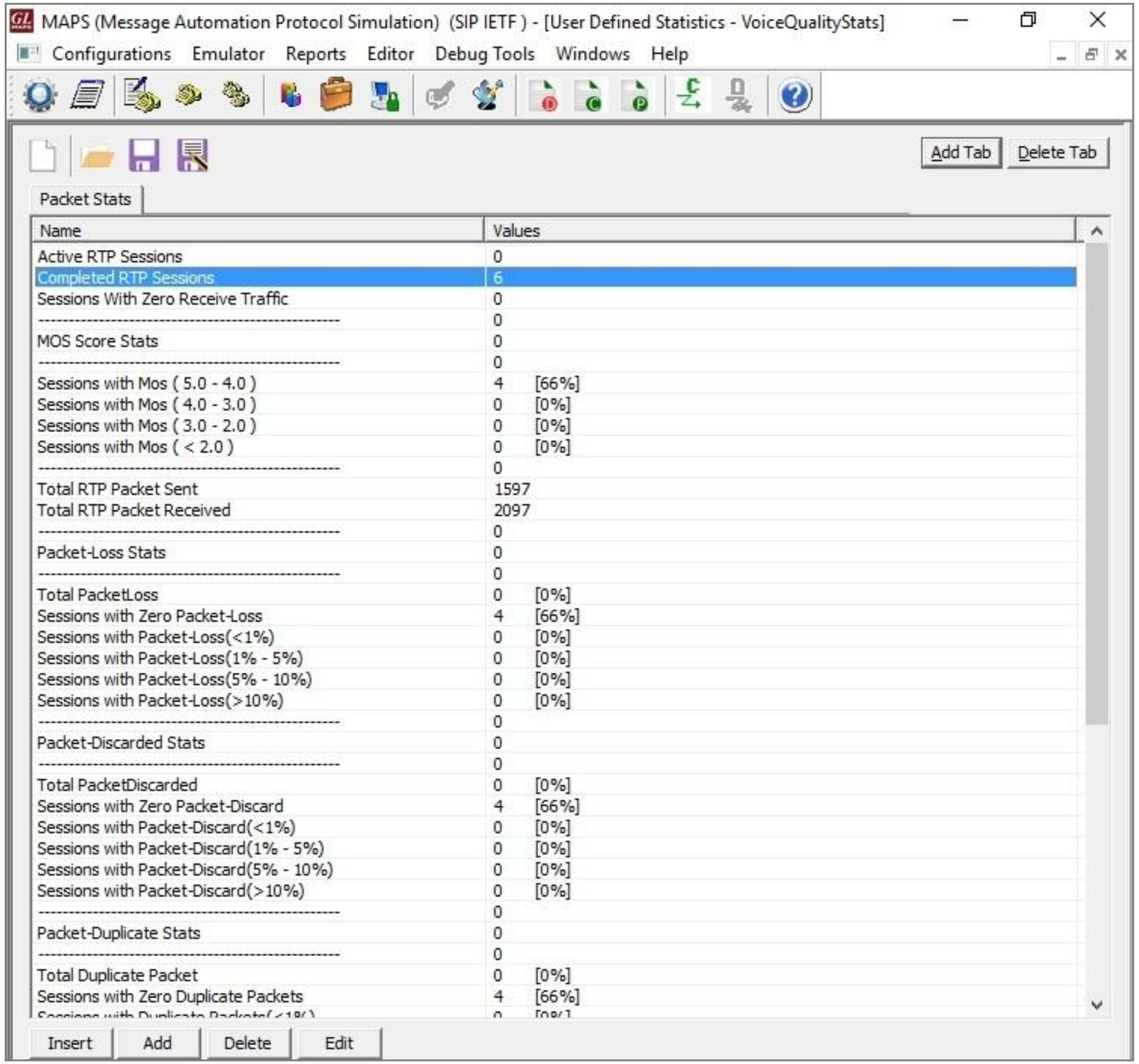
GL Communications

Call Generation with IVR Traffic



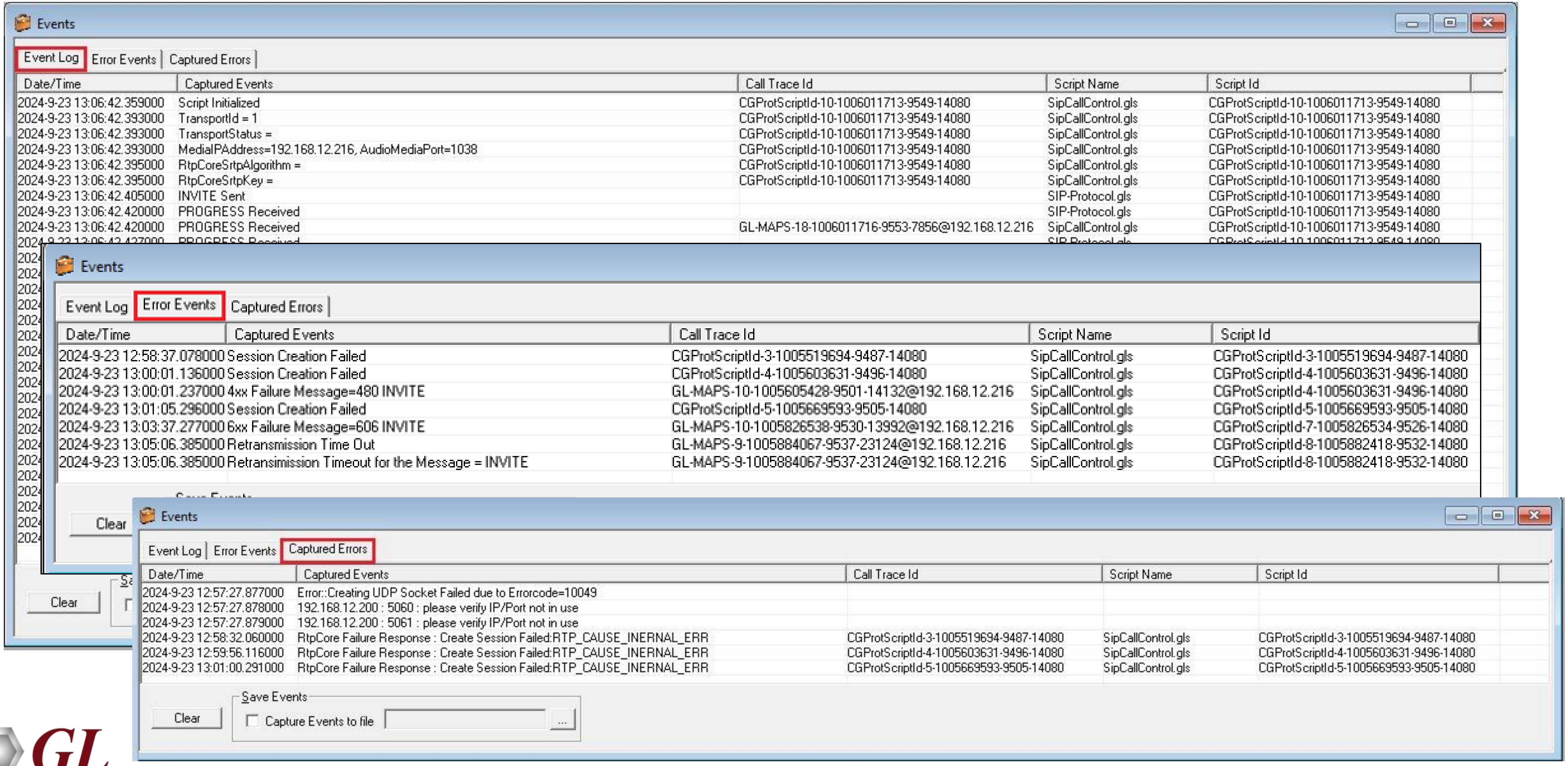
RTP Voice Quality Measurements

- RTP based Voice Quality (MOS and R-Factor) measurement are calculated and updated periodically for the received streams
- Call quality metrics includes Listening MOS, Conversational MOS, Packet Loss, Discarded Packets, Out of Sequence Packets, Duplicate Packets, Delay and Jitter



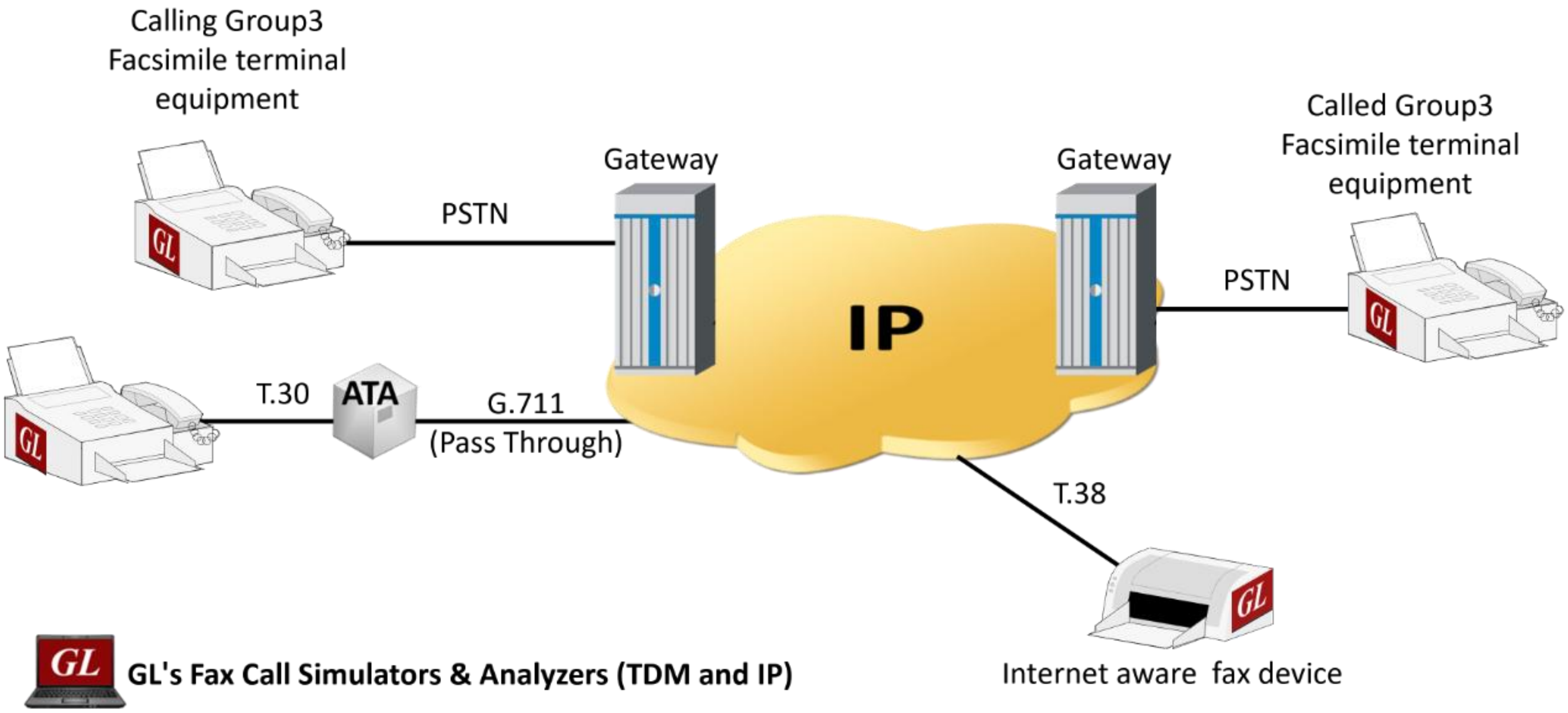
GL Communications

Event Log, Error Events, Captured Errors



GL Communications

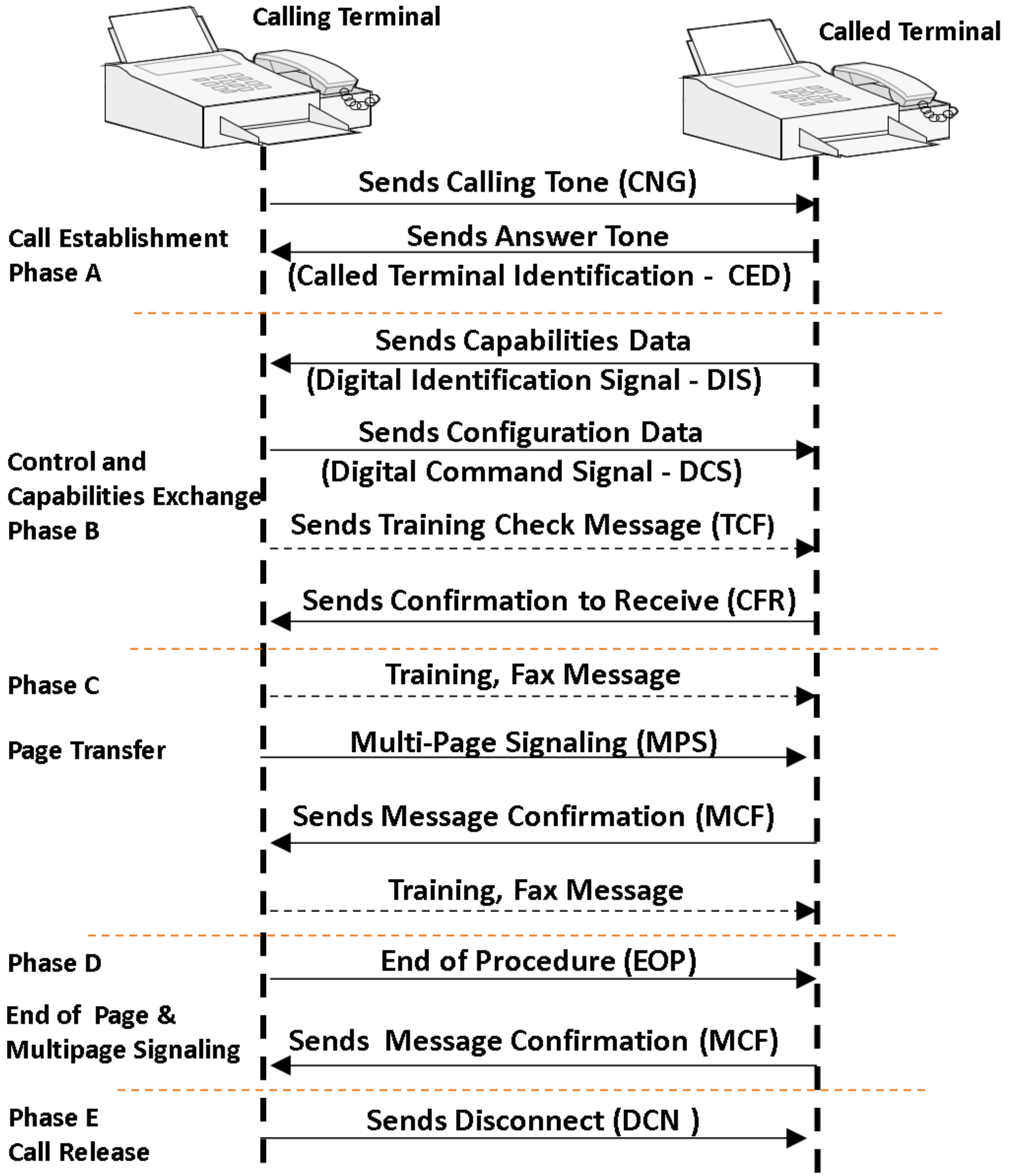
Fax Simulation over IP



- RTP G.711 Pass Through Fax Simulation (PKS200)
- T.38 Fax Simulation over UDPTL (PKS211)

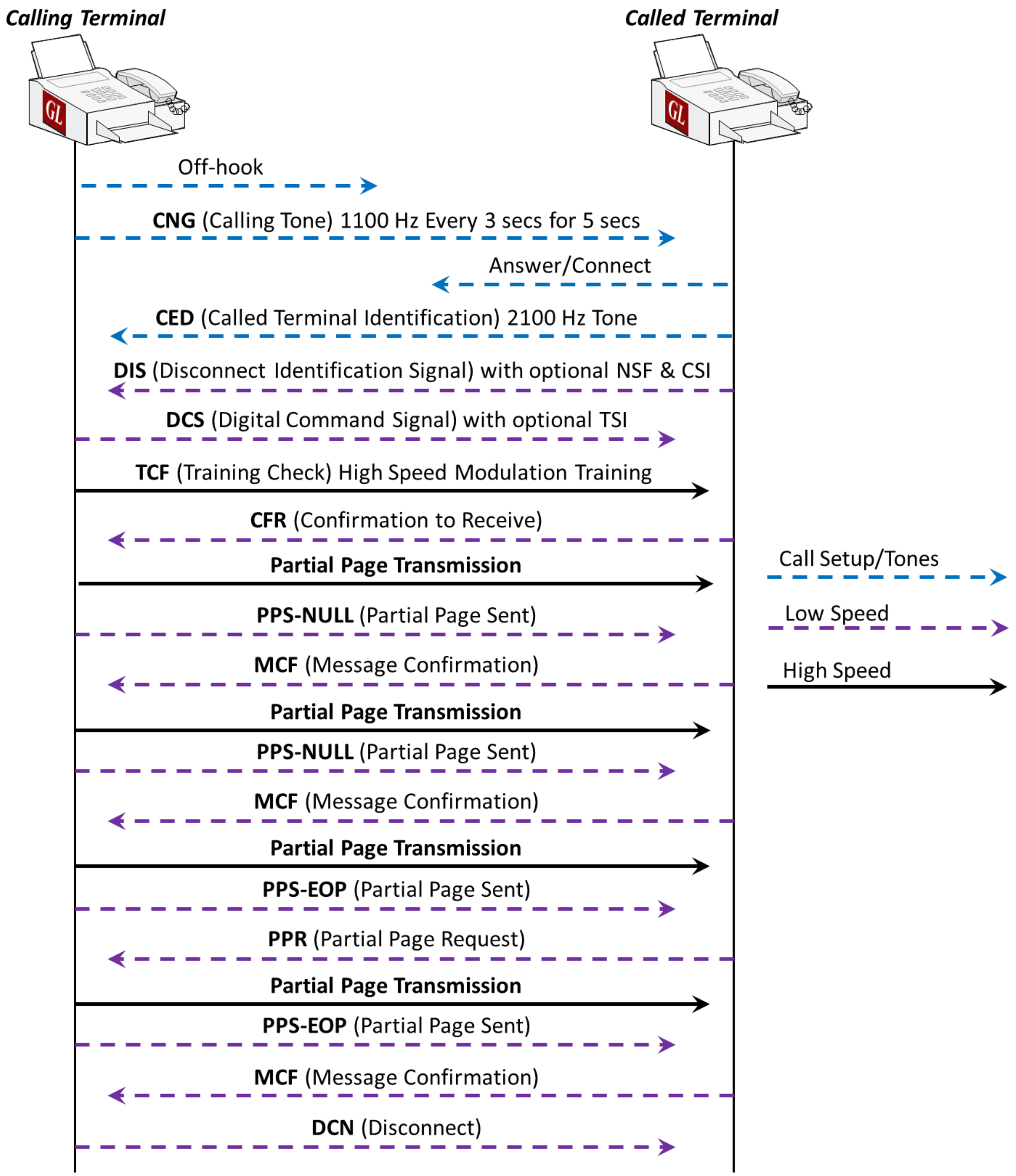
GL Communications

Call Scenarios - Fax T.30

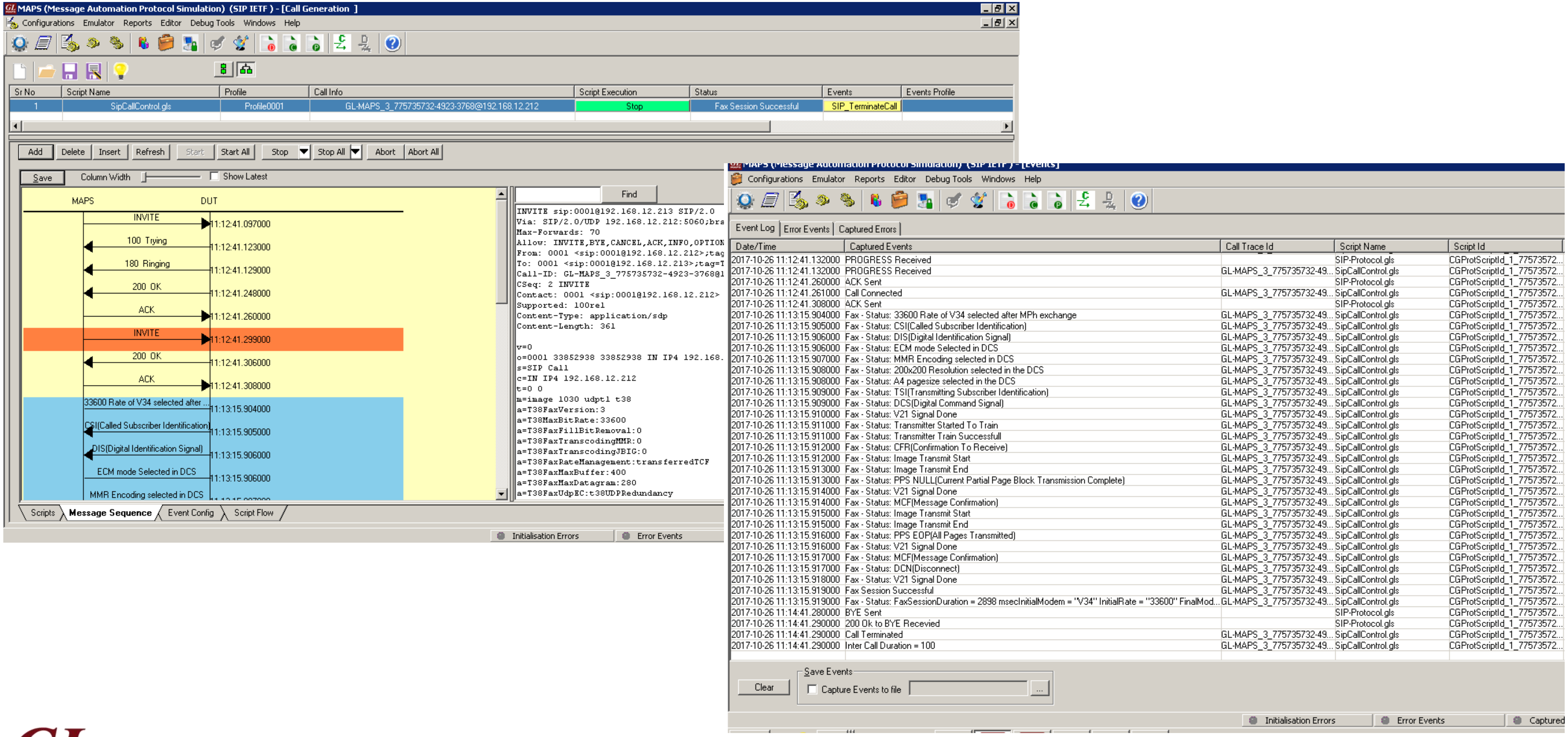


GL Communications

T.38 Fax Emulation over IP using MAPS™

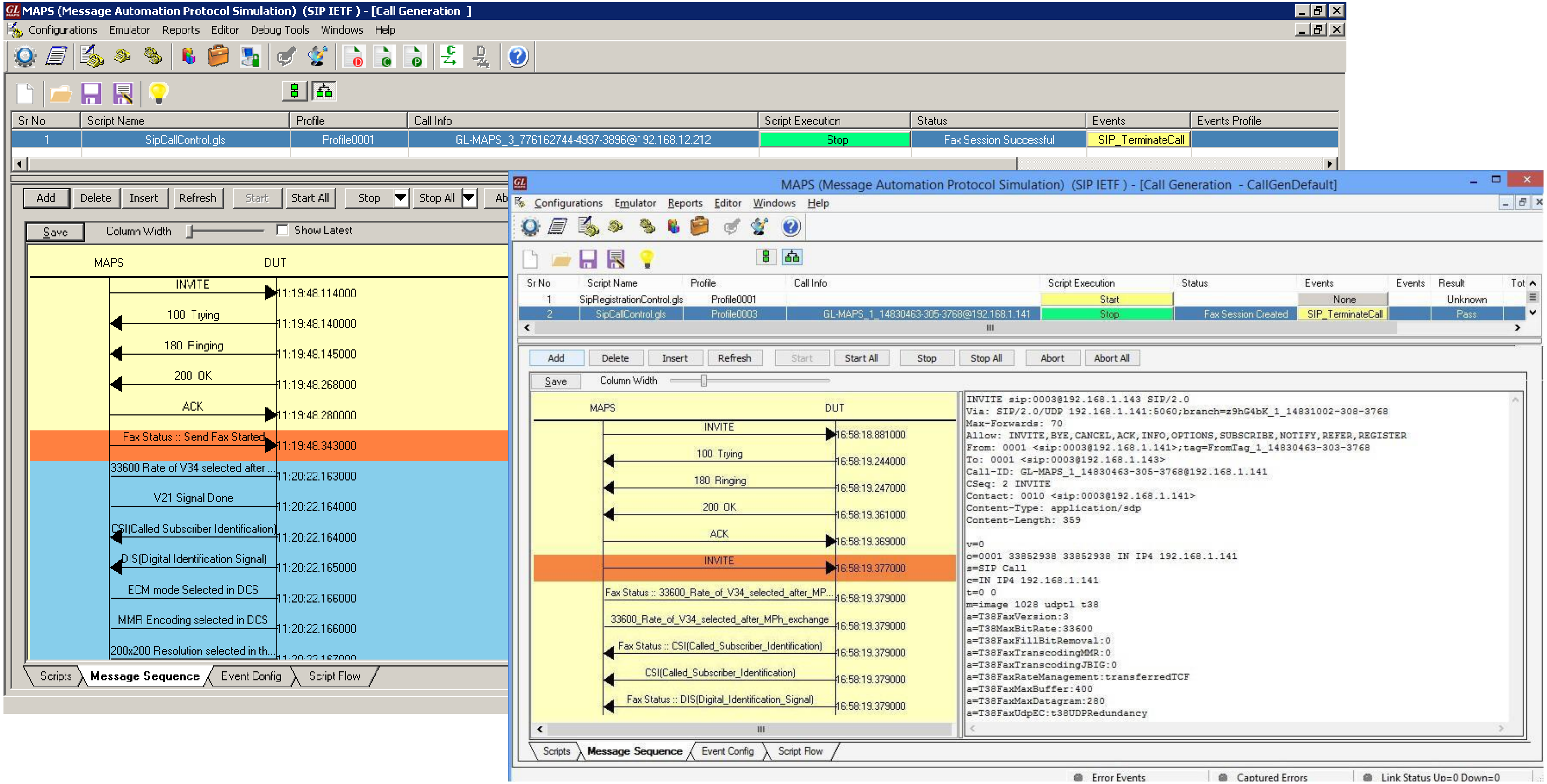


T.38 Fax Call in Progress and Related Events



GL Communications

Call Generation with FAX Traffic



GL Communications

FAX Traffic Events

Events

Event Log | Error Events | Captured Errors

Date/Time	Captured Events	Call Trace Id	Script Name	Script Id
2015-1-15 15:27:08.544000	UDP Port = 5060		SIP-Protocol.gls	CGProtScriptId_14_187697168-4524-6432
2015-1-15 15:27:08.544000	INVITE Sent		SIP-Protocol.gls	CGProtScriptId_14_187697168-4524-6432
2015-1-15 15:27:08.556000	PROGRESS Received	GL-MAPS_1_187697169-4528-8320@192.168.1.203	SipCallControl.gls	CGProtScriptId_14_187697168-4524-6432
2015-1-15 15:27:08.556000	PROGRESS Received		SIP-Protocol.gls	CGProtScriptId_14_187697168-4524-6432
2015-1-15 15:27:08.556000	PROGRESS Received	GL-MAPS_1_187697169-4528-8320@192.168.1.203	SipCallControl.gls	CGProtScriptId_14_187697168-4524-6432
2015-1-15 15:27:08.673000	ACK Sent		SIP-Protocol.gls	CGProtScriptId_14_187697168-4524-6432
2015-1-15 15:27:08.673000	Call Connected	GL-MAPS_1_187697169-4528-8320@192.168.1.203	SipCallControl.gls	CGProtScriptId_14_187697168-4524-6432
2015-1-15 15:27:08.678000	Sending RTP Fax	GL-MAPS_1_187697169-4528-8320@192.168.1.203	SipCallControl.gls	CGProtScriptId_14_187697168-4524-6432
2015-1-15 15:27:32.397000	RTP Fax Sent	GL-MAPS_1_187697169-4528-8320@192.168.1.203	SipCallControl.gls	CGProtScriptId_14_187697168-4524-6432
2015-1-15 15:27:39.386000	BYE Sent		SIP-Protocol.gls	CGProtScriptId_14_187697168-4524-6432
2015-1-15 15:27:39.394000	200 Ok to BYE Recevied		SIP-Protocol.gls	CGProtScriptId_14_187697168-4524-6432
2015-1-15 15:27:39.394000	Call Terminated	GL-MAPS_1_187697169-4528-8320@192.168.1.203	SipCallControl.gls	CGProtScriptId_14_187697168-4524-6432
2015-1-15 15:27:39.394000	Inter Call Duration = 1000	GL-MAPS_1_187697169-4528-8320@192.168.1.203	SipCallControl.gls	CGProtScriptId_14_187697168-4524-6432

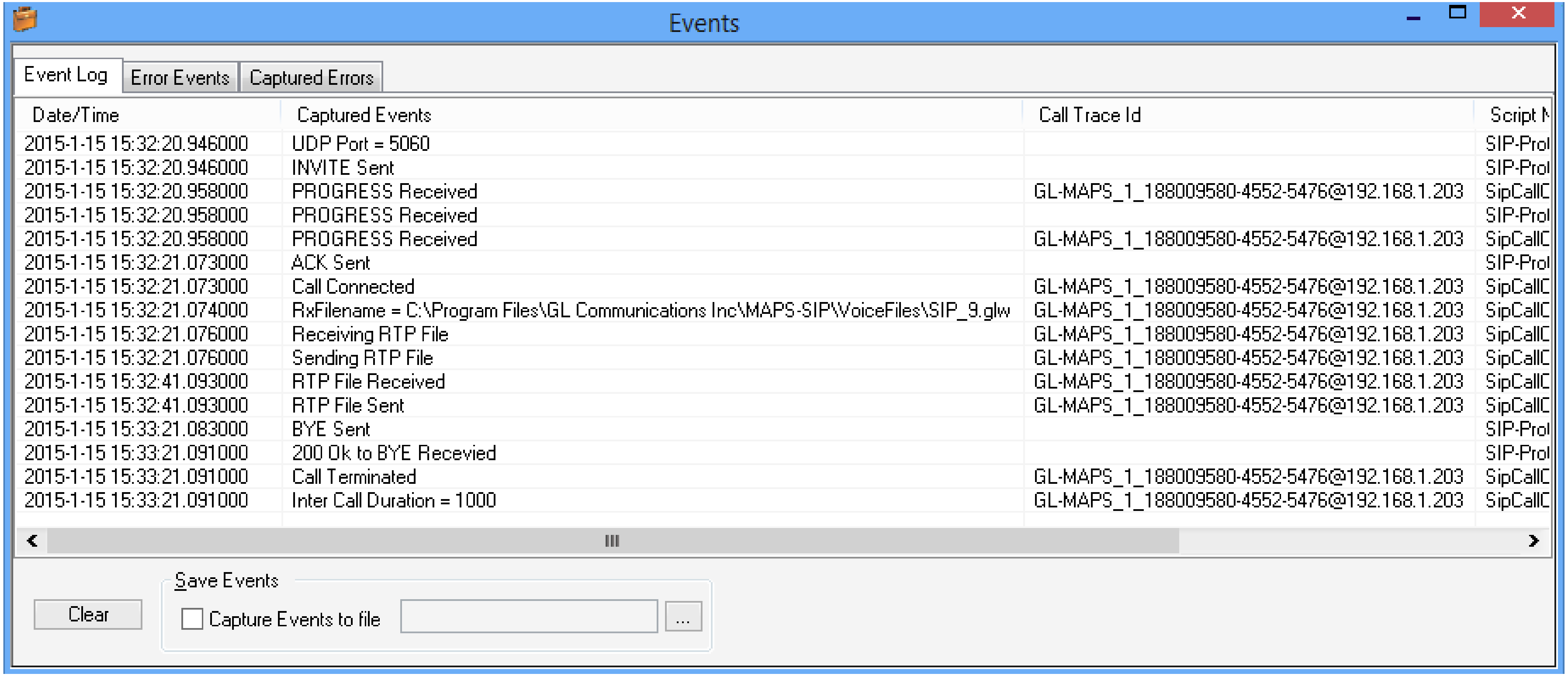
Clear

Save Events

Capture Events to file ...

GL Communications

File Traffic Events



Events

Event Log | Error Events | Captured Errors

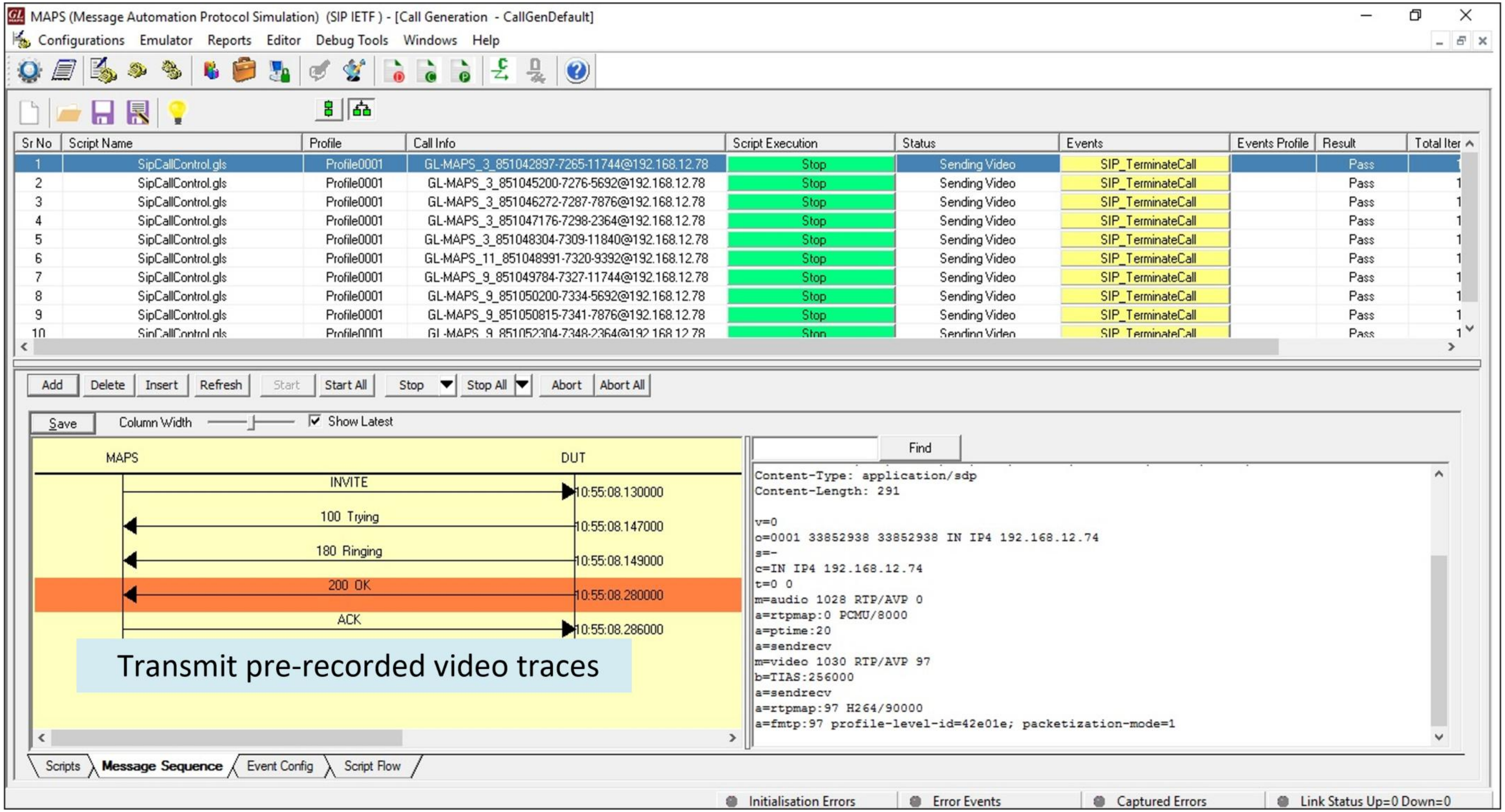
Date/Time	Captured Events	Call Trace Id	Script N
2015-1-15 15:32:20.946000	UDP Port = 5060		SIP-Prol
2015-1-15 15:32:20.946000	INVITE Sent		SIP-Prol
2015-1-15 15:32:20.958000	PROGRESS Received	GL-MAPS_1_188009580-4552-5476@192.168.1.203	SipCallC
2015-1-15 15:32:20.958000	PROGRESS Received		SIP-Prol
2015-1-15 15:32:20.958000	PROGRESS Received	GL-MAPS_1_188009580-4552-5476@192.168.1.203	SipCallC
2015-1-15 15:32:21.073000	ACK Sent		SIP-Prol
2015-1-15 15:32:21.073000	Call Connected	GL-MAPS_1_188009580-4552-5476@192.168.1.203	SipCallC
2015-1-15 15:32:21.074000	RxFilename = C:\Program Files\GL Communications Inc\MAPS-SIP\VoiceFiles\SIP_9.glw	GL-MAPS_1_188009580-4552-5476@192.168.1.203	SipCallC
2015-1-15 15:32:21.076000	Receiving RTP File	GL-MAPS_1_188009580-4552-5476@192.168.1.203	SipCallC
2015-1-15 15:32:21.076000	Sending RTP File	GL-MAPS_1_188009580-4552-5476@192.168.1.203	SipCallC
2015-1-15 15:32:41.093000	RTP File Received	GL-MAPS_1_188009580-4552-5476@192.168.1.203	SipCallC
2015-1-15 15:32:41.093000	RTP File Sent	GL-MAPS_1_188009580-4552-5476@192.168.1.203	SipCallC
2015-1-15 15:33:21.083000	BYE Sent		SIP-Prol
2015-1-15 15:33:21.091000	200 Ok to BYE Recevied		SIP-Prol
2015-1-15 15:33:21.091000	Call Terminated	GL-MAPS_1_188009580-4552-5476@192.168.1.203	SipCallC
2015-1-15 15:33:21.091000	Inter Call Duration = 1000	GL-MAPS_1_188009580-4552-5476@192.168.1.203	SipCallC

Clear

Save Events

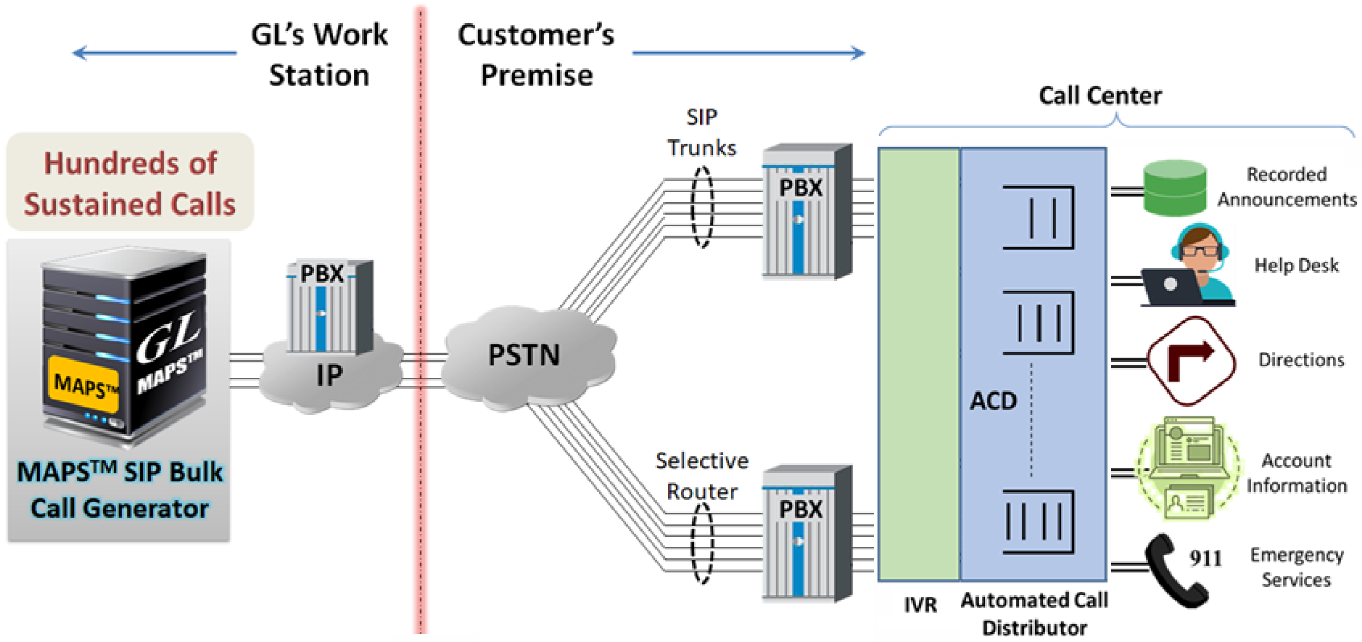
Capture Events to file ...

Video Call Generation

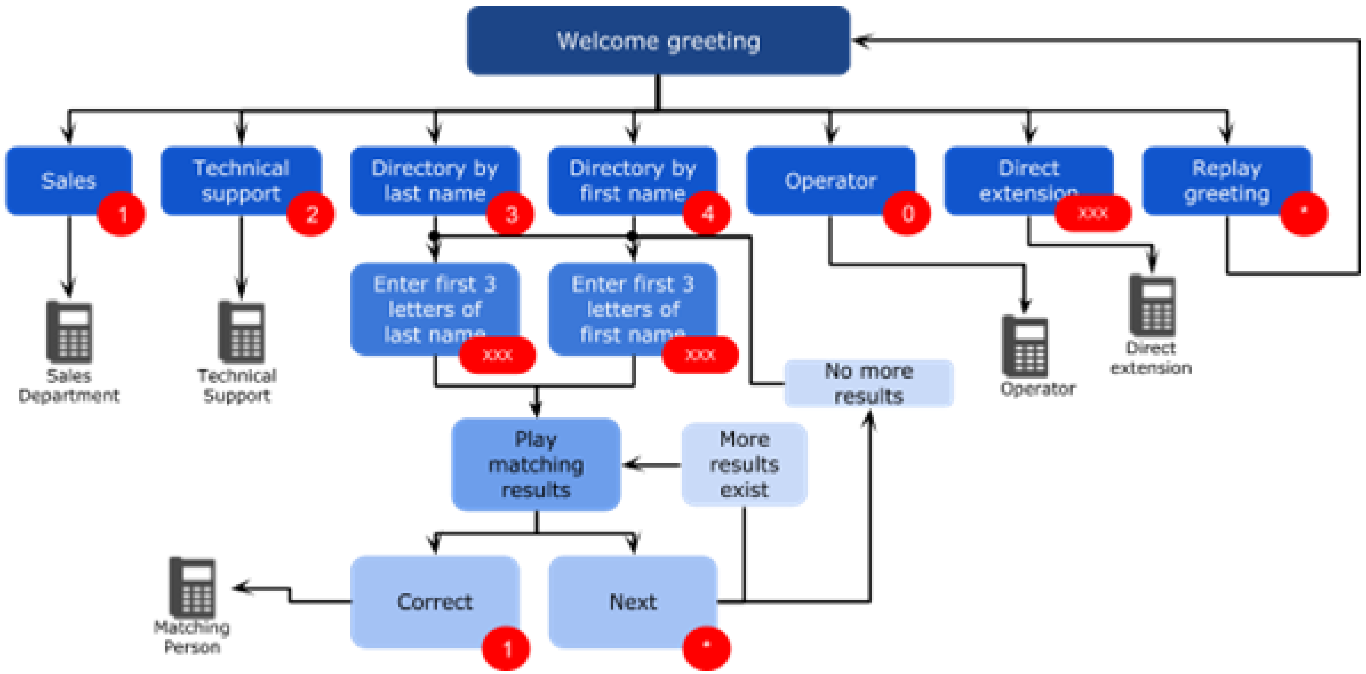


Speech to Text Interactive Voice Response (IVR)

- MAPS™ SIP with GL's Speech Transcription Server provides automated IVR testing by using speech to text to navigate through an IVR tree
- IVR prompts are recorded by MAPS™ SIP and transcribed by the Speech Transcription Server
- Transcribed text is compared to an expected text at each IVR stage to confirm the prompt
- Once the IVR prompt is confirmed, MAPS™ sends DTMF or voice-based responses to move to the next stage
- The expected IVR prompts and responses are defined by the customer to ensure completely customizable tests that are suitable for all IVR systems



GL's Interactive Voice Response Scenario

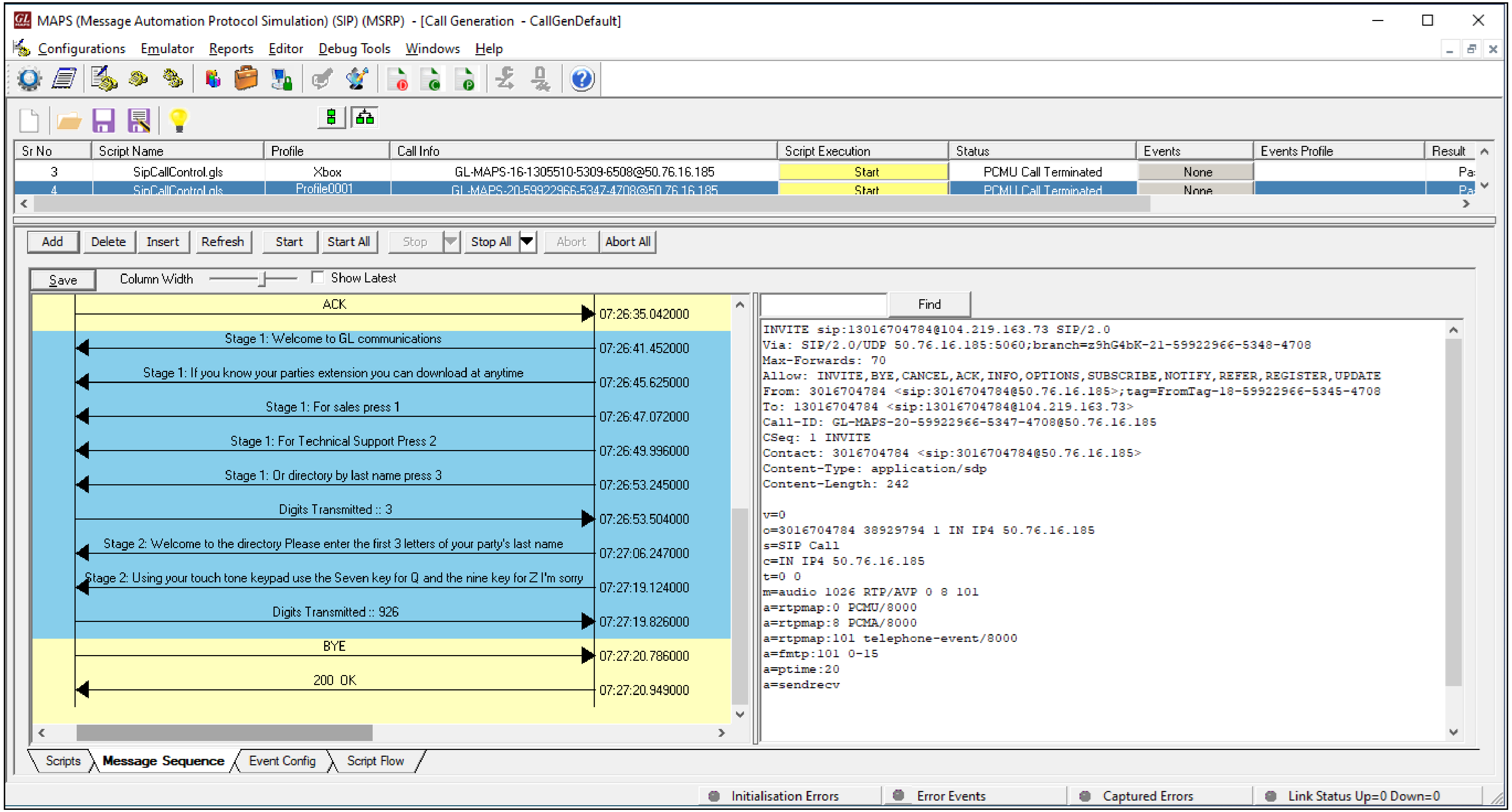


- The CSV file in the screenshot below shows a basic IVR traversal test of this IVR system

	A	B	C	D	E	F	G
1	IVRIndex	IVRPromptLanguage	IVRExpectedTranscript	IVRResponseType	IVRResponseDTMF	IVRResponseSpeech	IVRNextPromptId
2	int	string	string	string	string	string	int
3	1	en-US	Welcome to GL Communications If you know your partys extension you can dial it at any time For sales press one for technical support press 2 for a directory by last name press 3	DTMF	3		2
4	2	en-US	Welcome to the directory. please enter the first 3 letters of your partys last name using your touch tone keypad Use the seven key for q and the nine key for z	DTMF	926		0

GL Communications

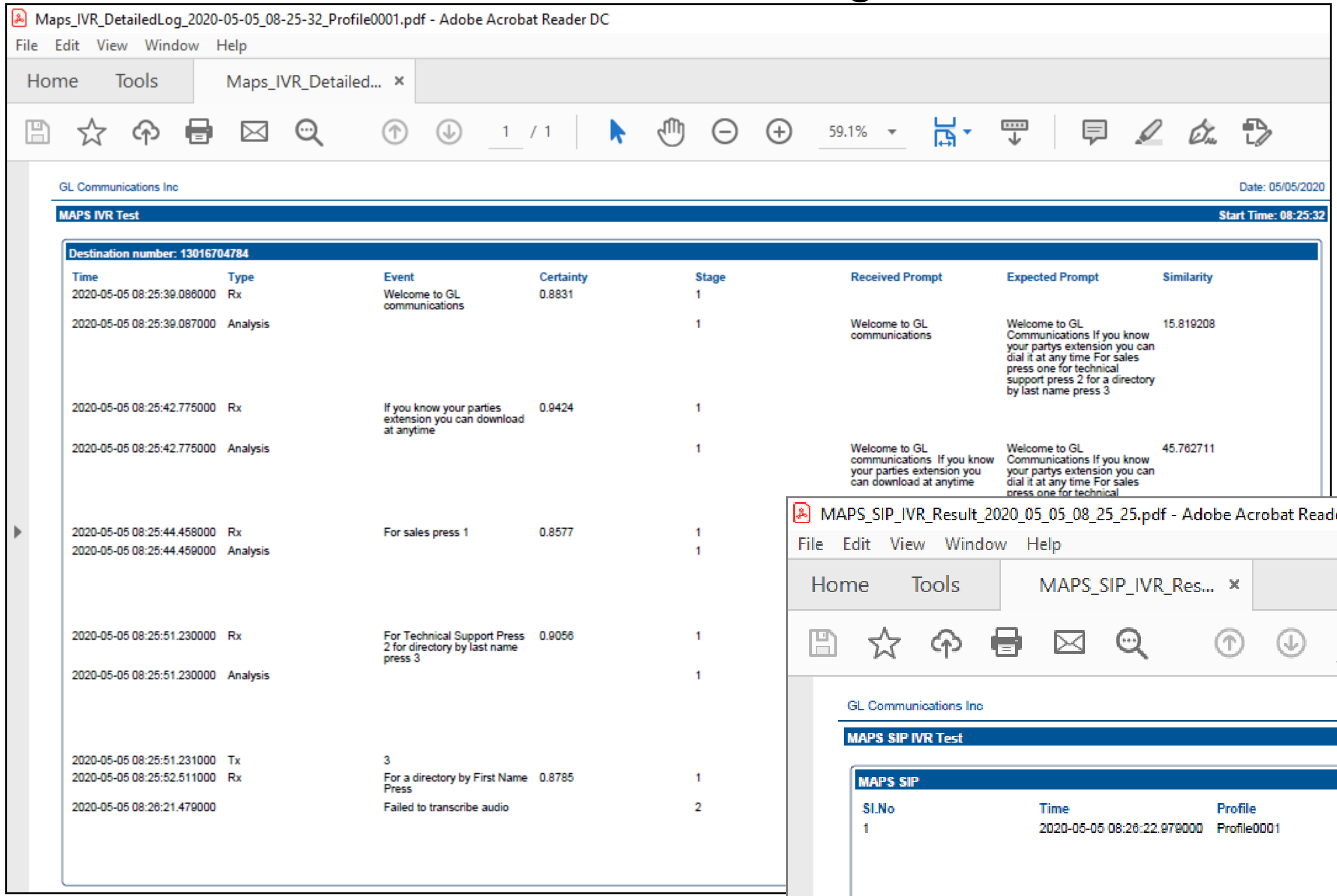
IVR Call Simulation



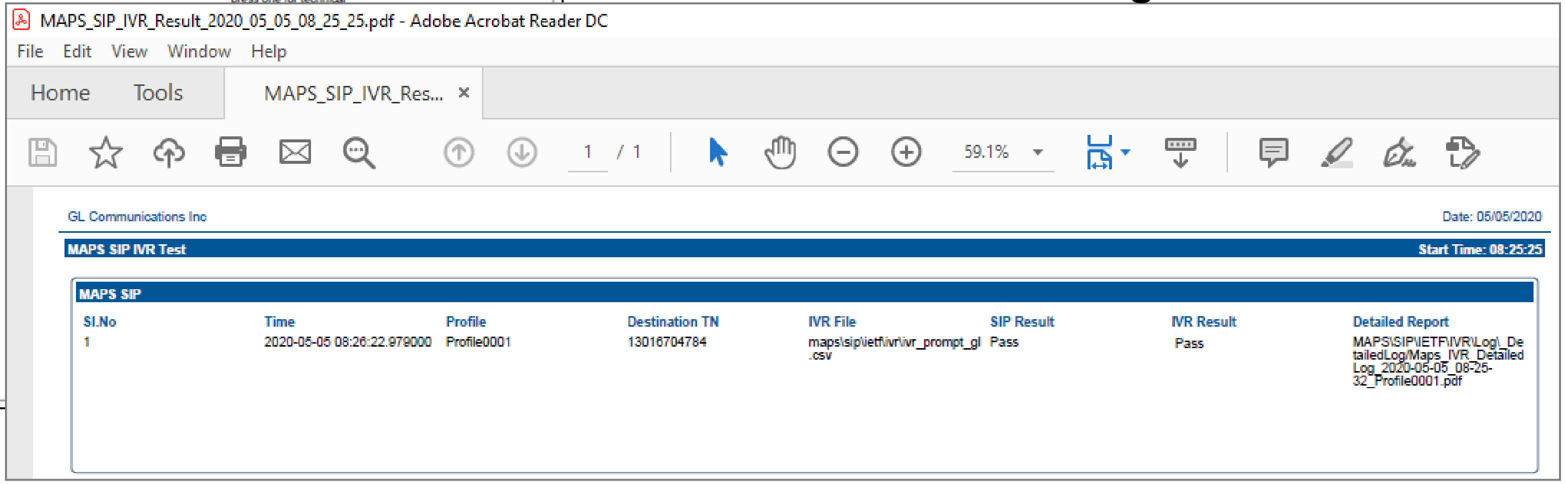
GL Communications

IVR Call Simulation Reports

SIP IVR Detailed Log

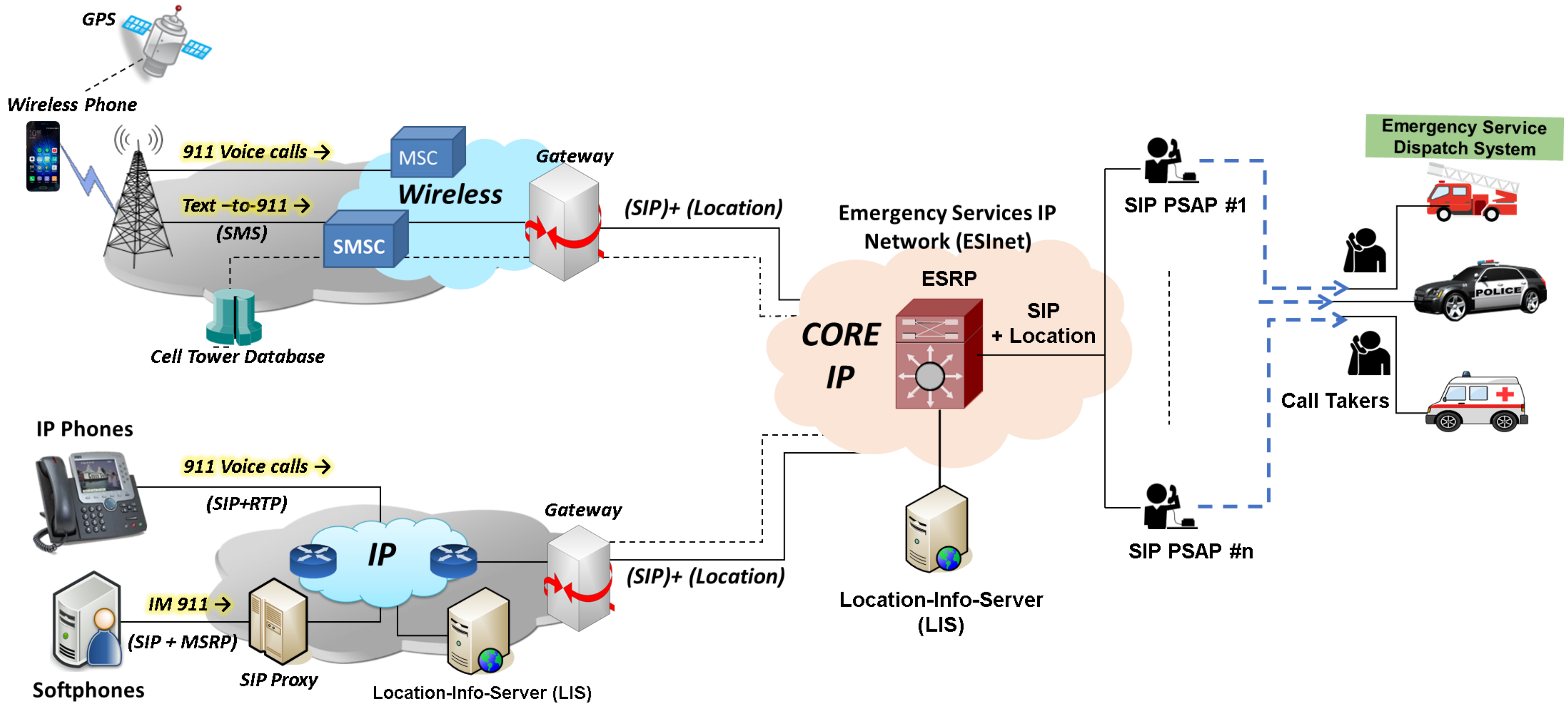


SIP IVR Result Log



Message Session Relay Protocol

Message Session Relay Protocol is a text-based, connection-oriented protocol for transmitting a series of related instant messages in the context of a session. MSRP sessions are typically arranged using SIP the same way a session of audio or video media is set up.



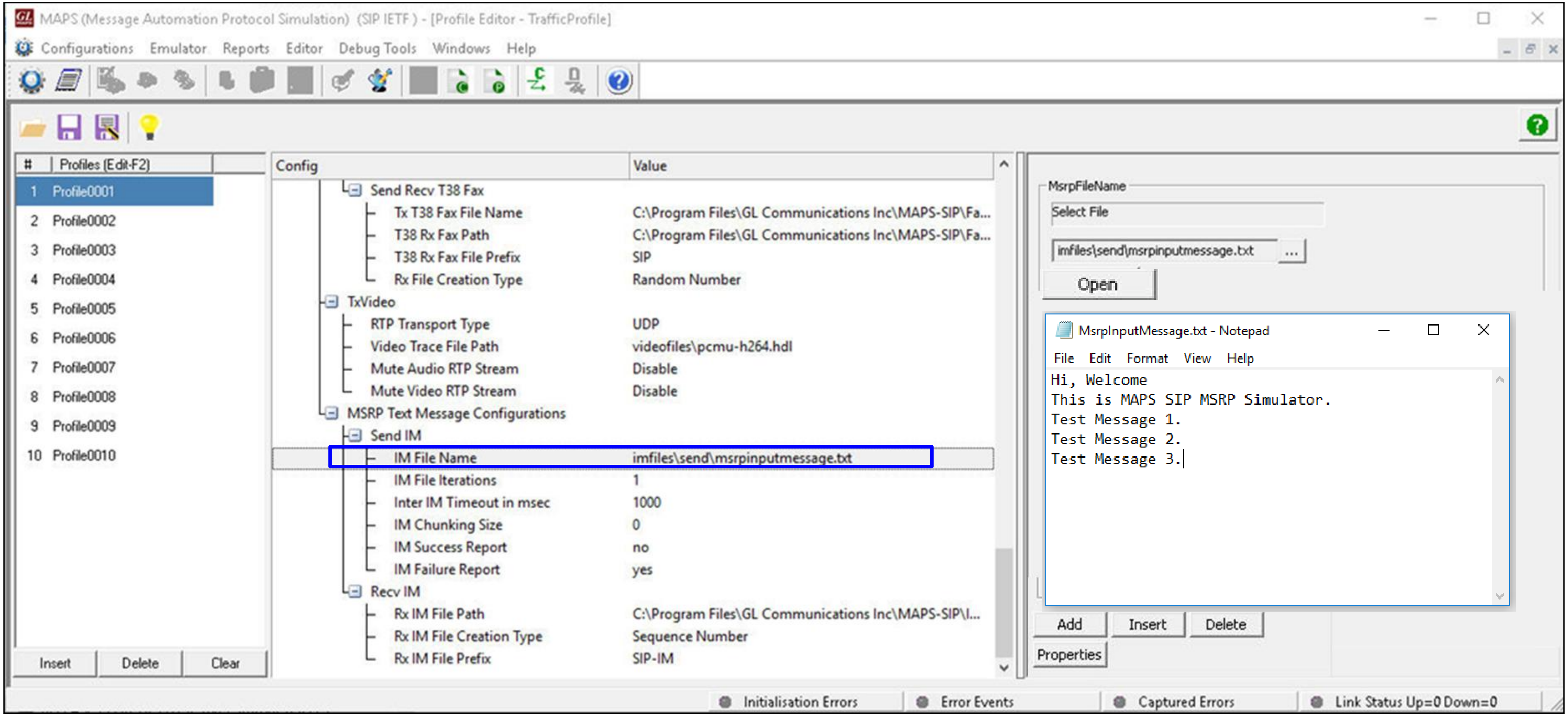
GL Communications

Message Session Relay Protocol (Contd.)

- Reads text messages from a pre-defined text file (user-configurable) and transmits them on established IM session
- Received messages on every MSRP session can be recorded to a text file
- Text file can have multiple lines of message. The CRLF will be the de-limiter to treat each line as a new message
- Supports message chunking with user configured chunk size
- Configuration options allow to:
  - ➢ Record and report success and failure reports in MSRP SEND method
  - ➢ Define message generation interval to control the message frequency on the call
- Supports mixed media SIP sessions i.e., Audio with IM / Video with IM / Only IM
- Provides IM statistics per call and aggregated statistics of over-all calls (Number and size of messages received and sent)
- Flexibility to validate MSRP devices through negative tests with invalid MSRP URI's, validate success and failure reports
- Supports up to 500 simultaneous MSRP sessions

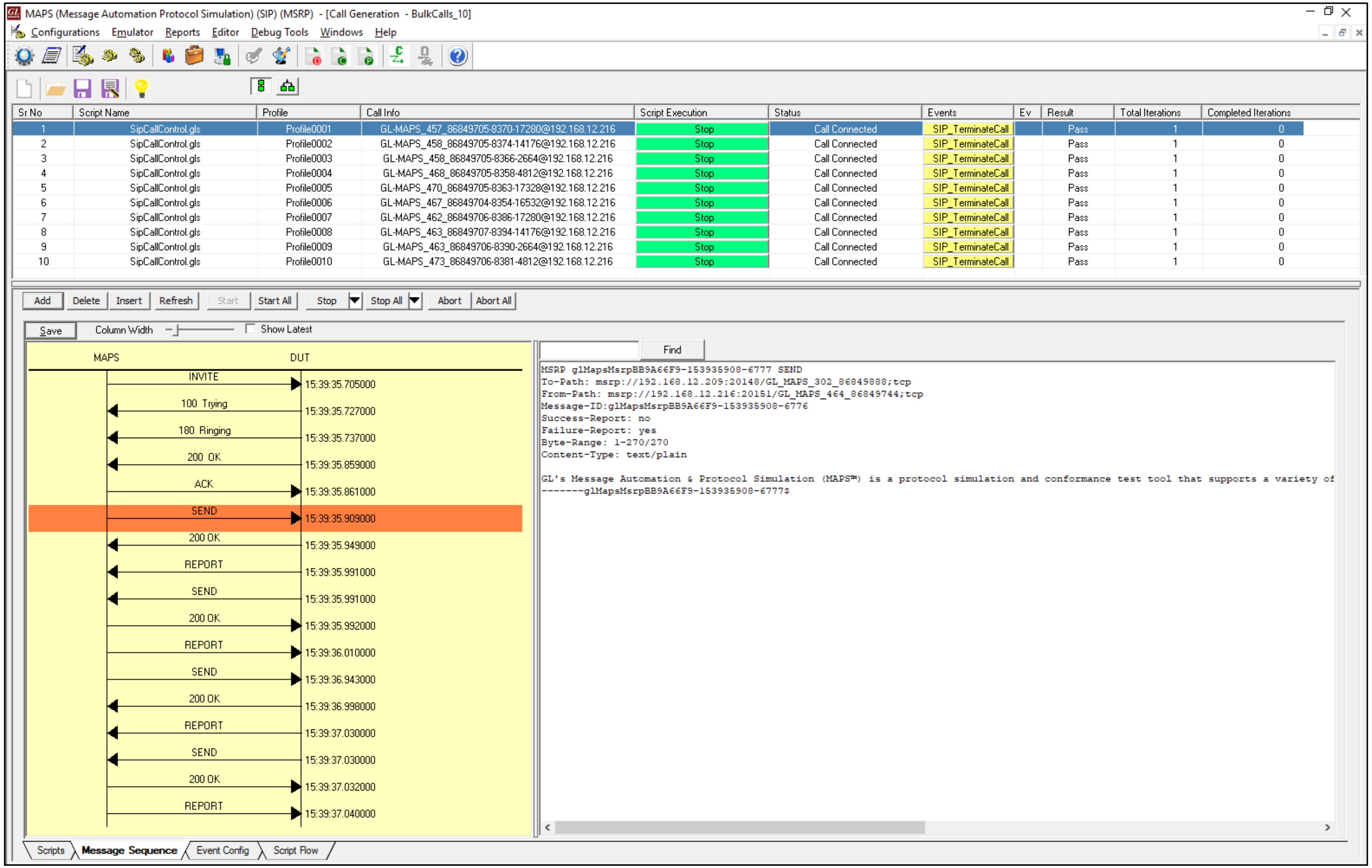
GL Communications

MSRP Traffic Configuration

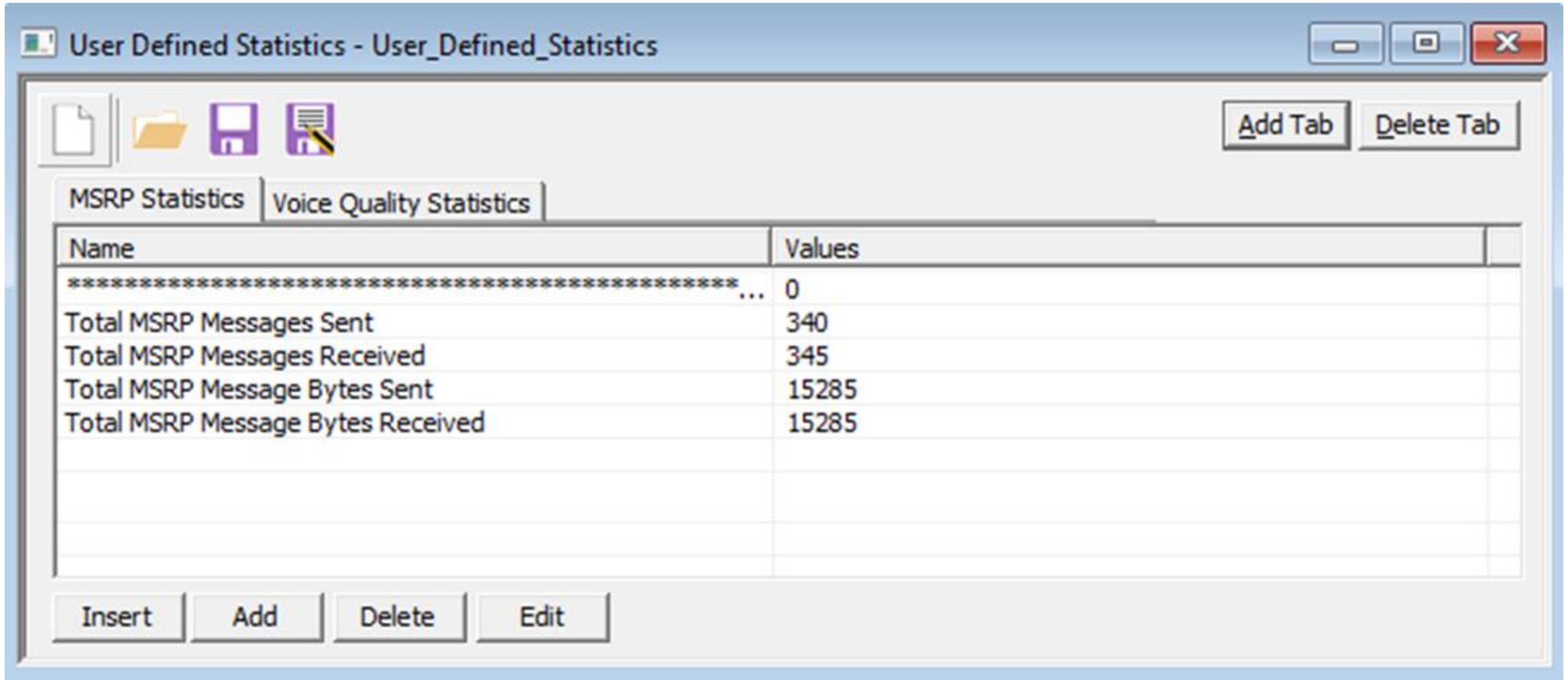


GL Communications

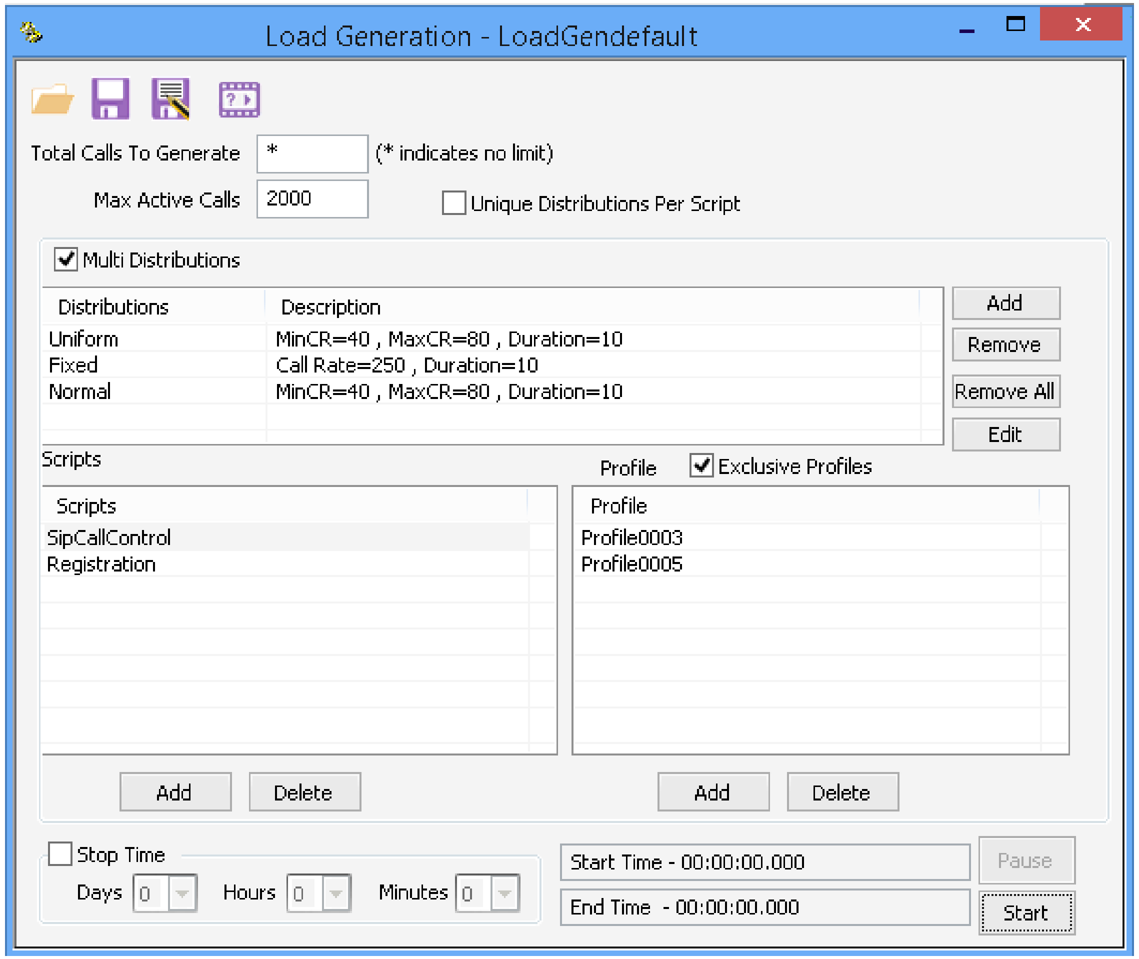
MSRP Call Generation



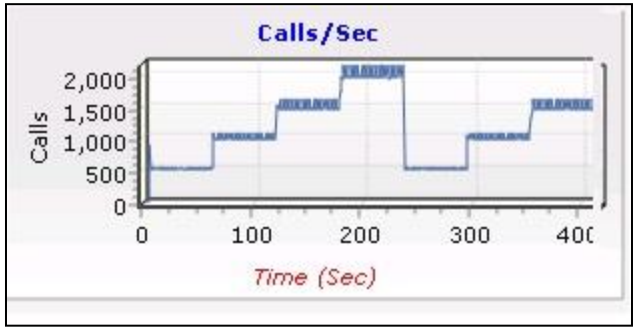
MSRP Statistics



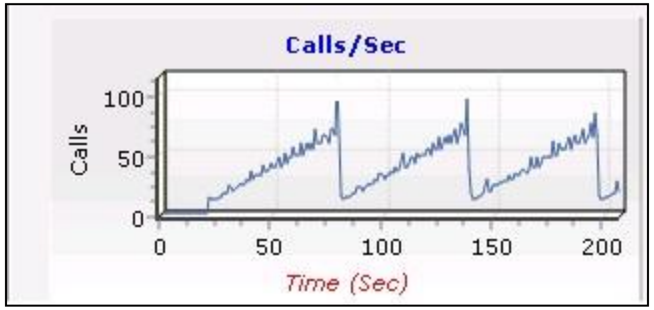
Load Generation



Step Statistical Distribution



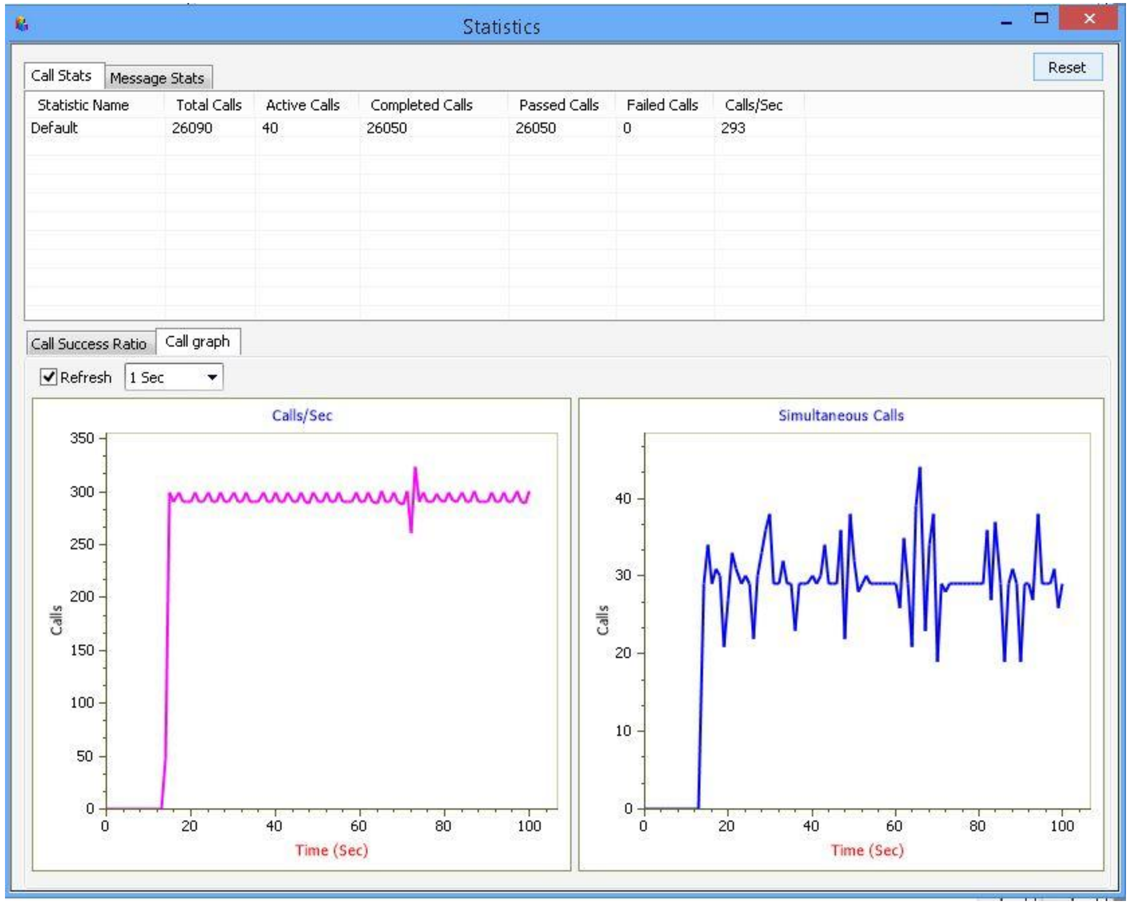
Ramp Statistical Distribution



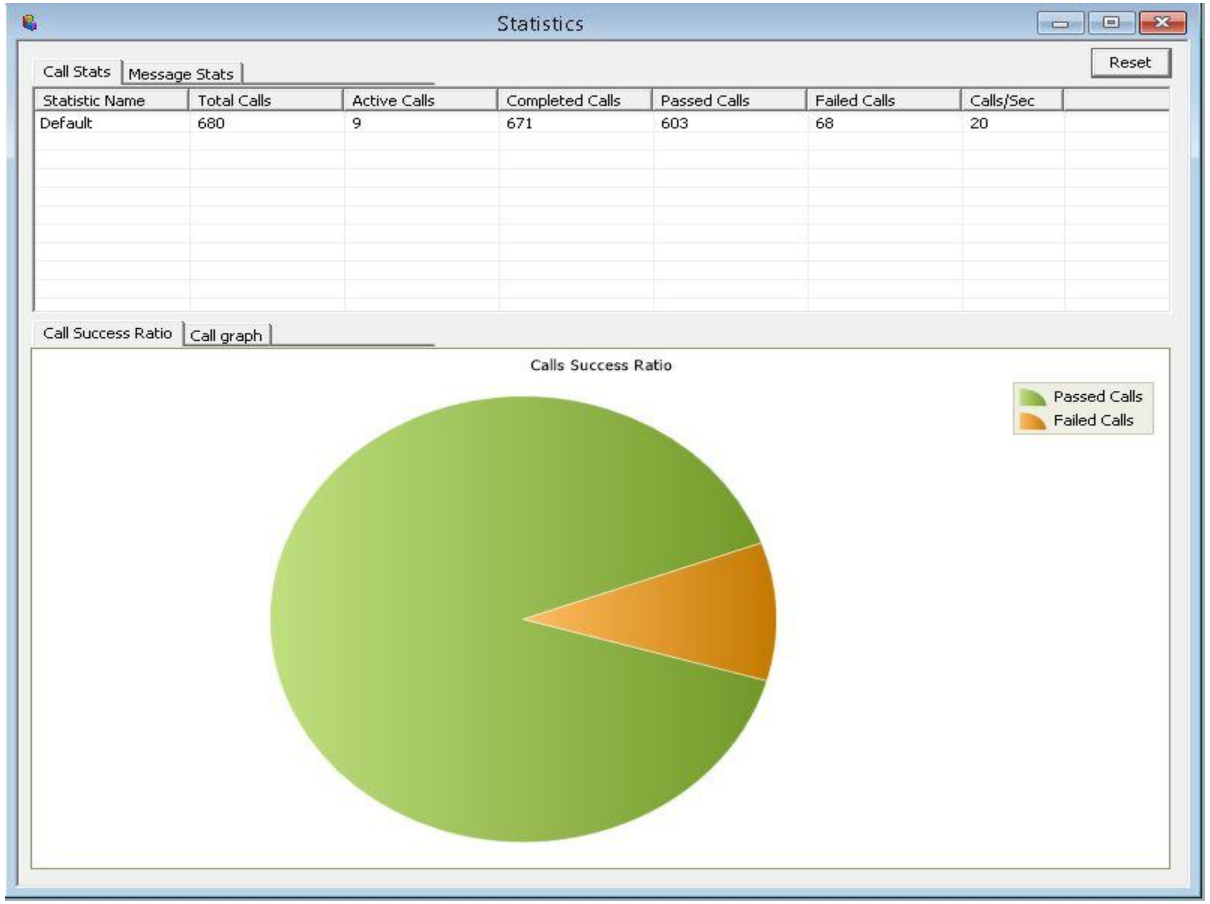
Saw-tooth Statistical Distribution

Success Call Ratio Statistics

Call Graph



Call Stats



Message Statistics

Statistics

Call Stats | Message Stats

Reset

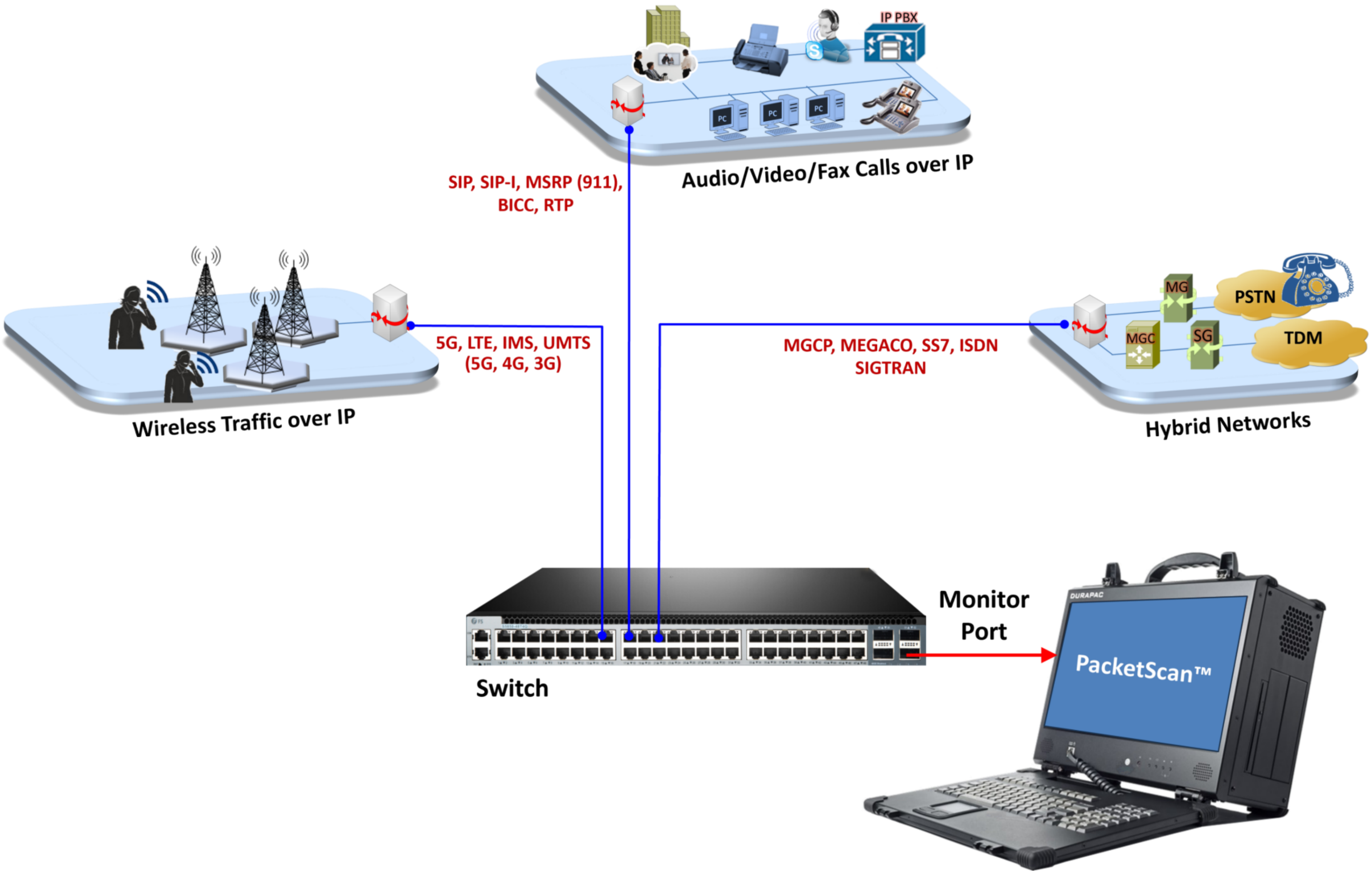
Message Type	Tx Count	Rx Count	Retransmit Count
100 INVITE	0	66040	0
180 INVITE	0	66040	0
200 BYE	0	46808	0
200 INVITE	0	66040	0
ACK	66040	0	0
BYE	46808	0	0
INVITE	66040	0	0

GL Communications

SIP RTP Analyzer - PacketScan™

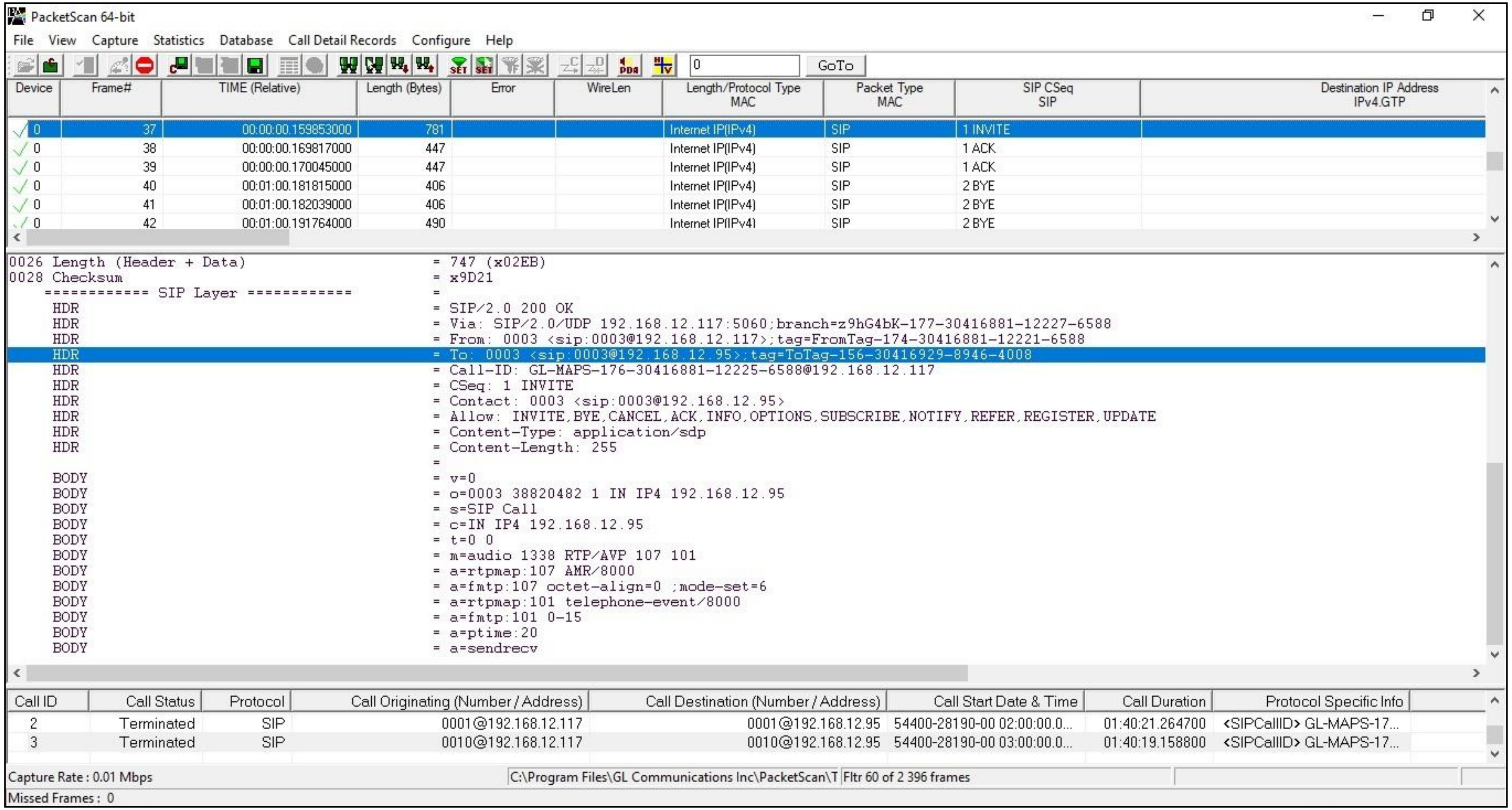
GL Communications

PacketScan™ VoIP Traffic Analysis
SIP / MSRP/ H323 / MEGACO / MGCP / RTP / RTCP / Video Analysis



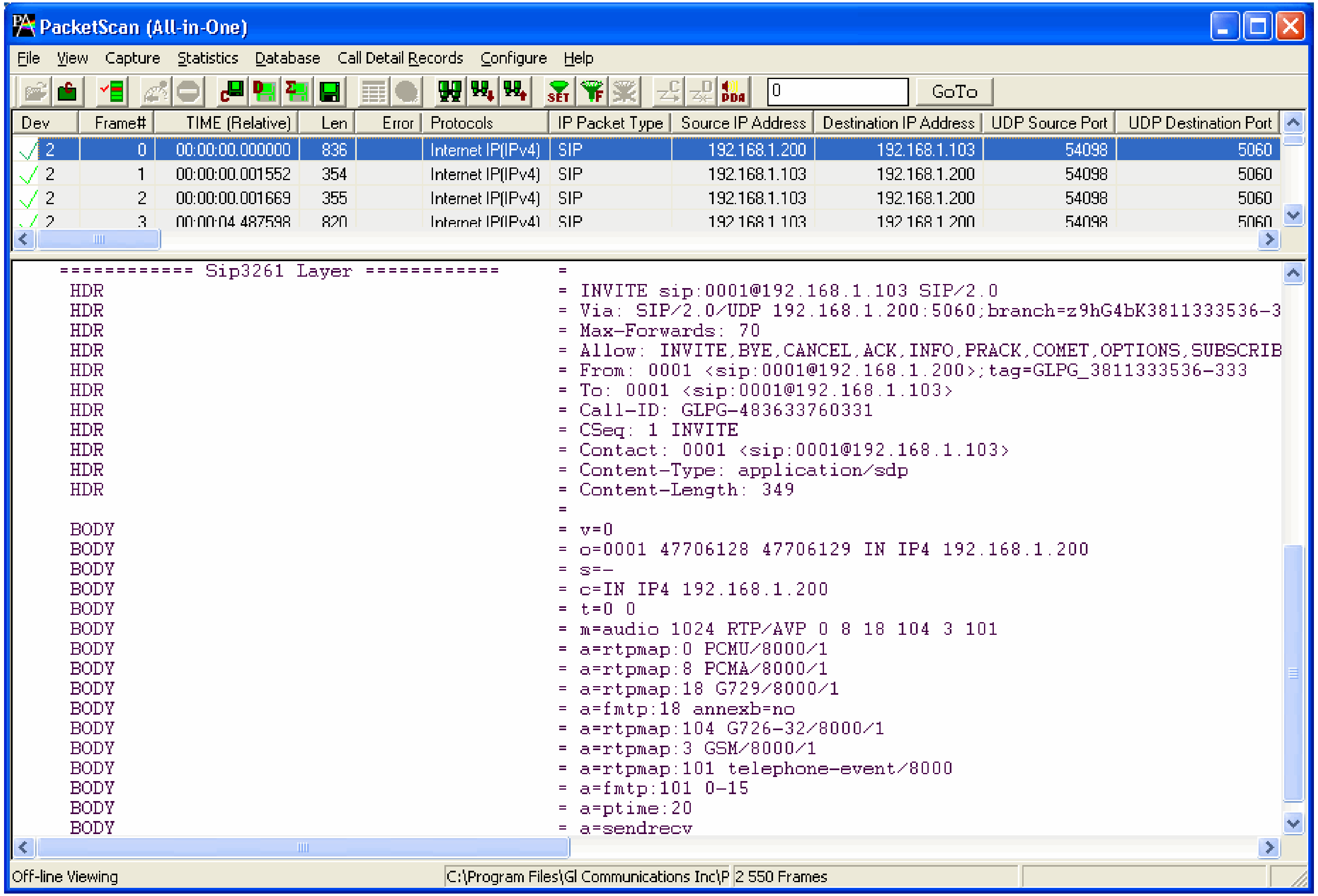
PacketScan™ Analyzer with SIP CDR

- Captures, segregates, and monitors packets; perform voice quality testing in real-time over VoIP network
- Unlimited traffic and signaling capturing capability; captured VoIP calls with video can be played back using 3rd party applications
- Can be deployed as a Probe for a centralized monitoring system with Oracle database



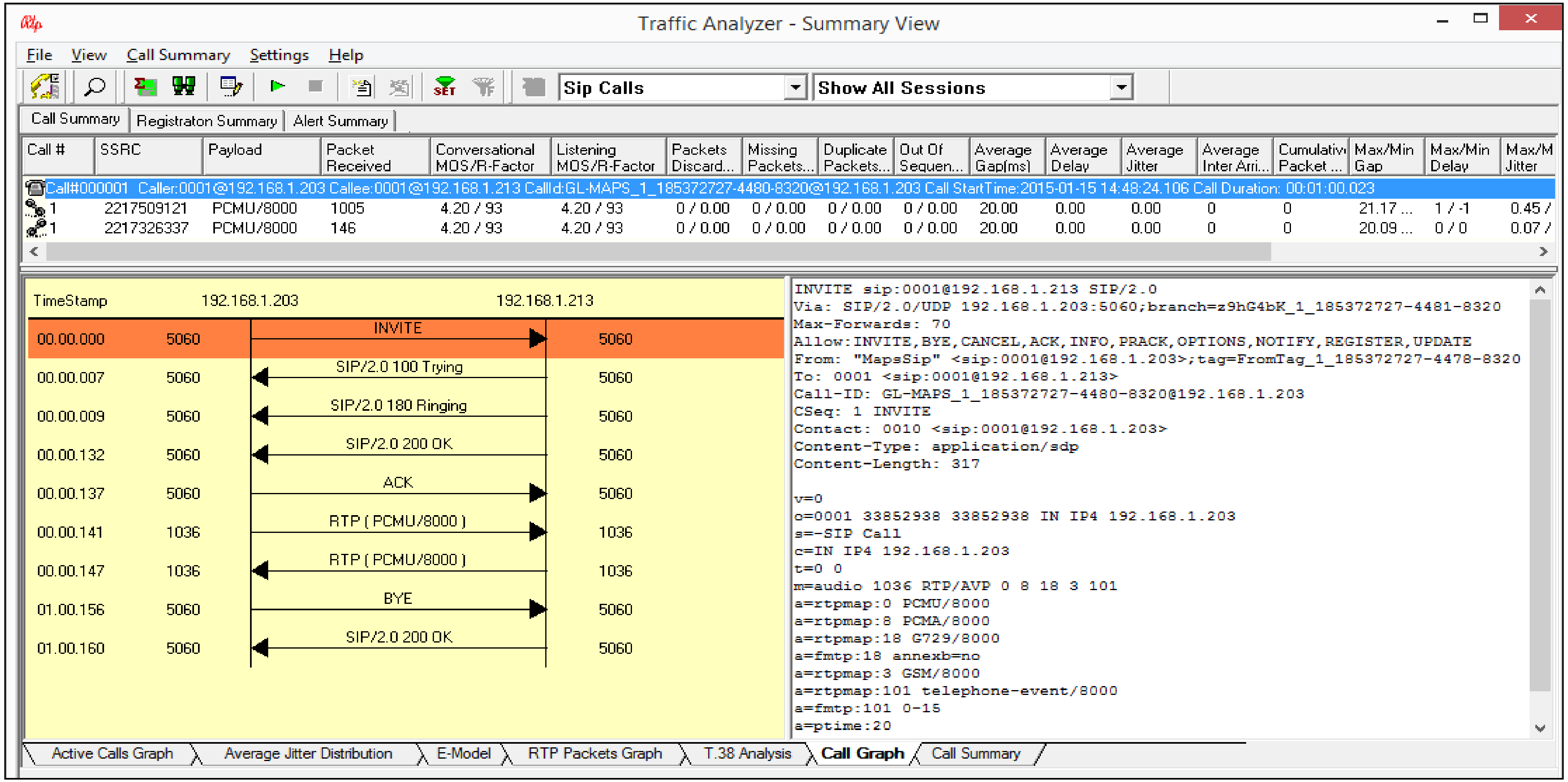
GL Communications

SIP Decode in PacketScan™



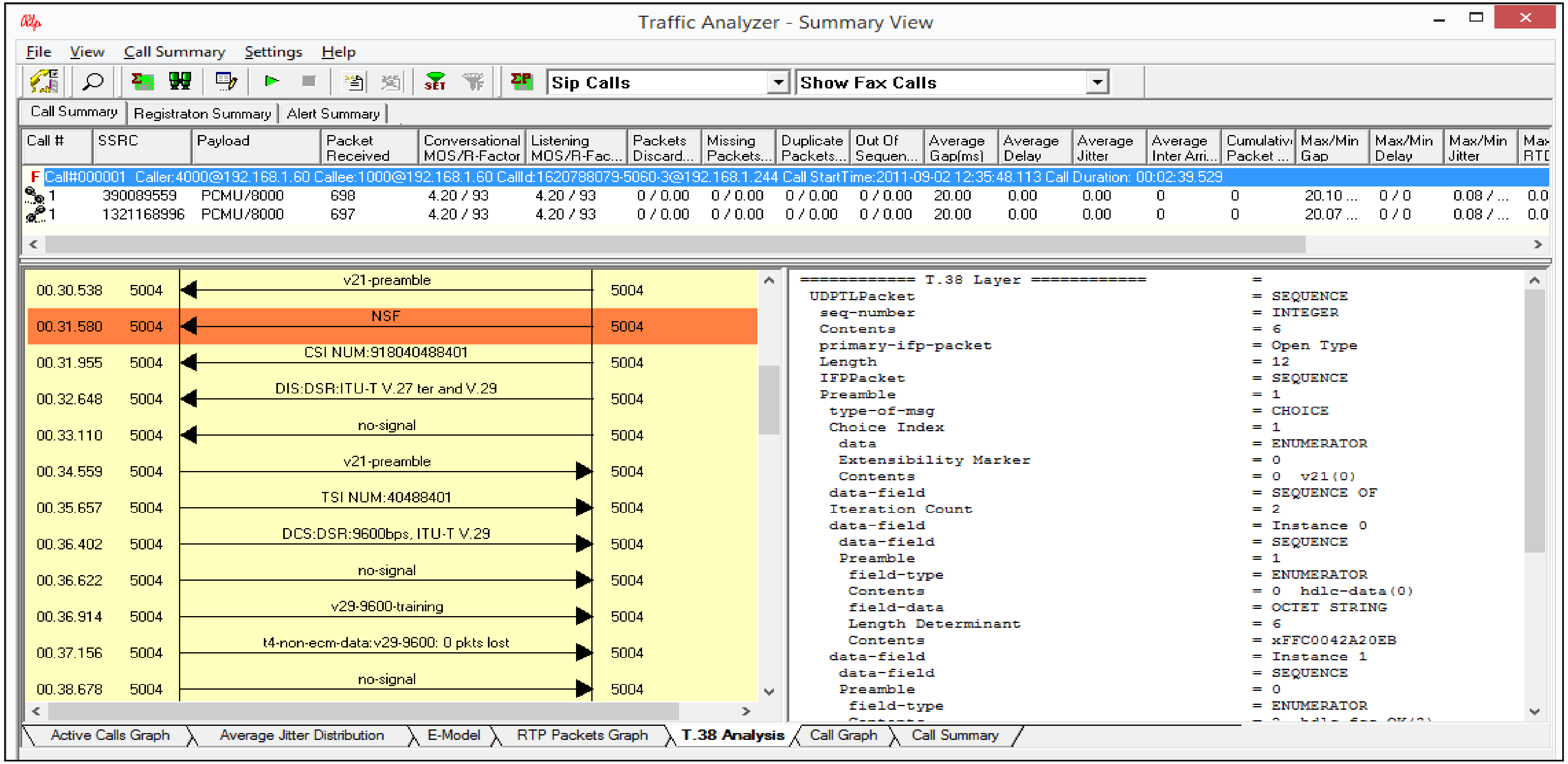
GL Communications

PacketScan™ PDA with SIP Call Summary

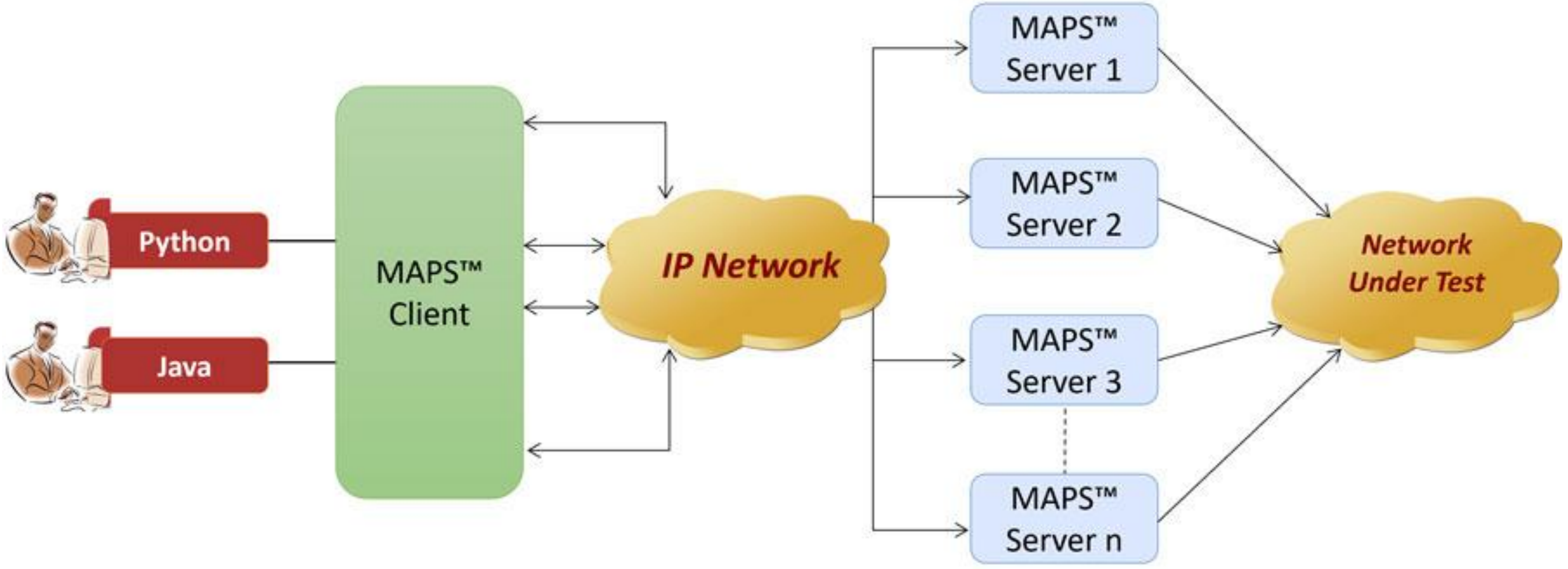


GL Communications

PacketScan™ Fax T.38 Analysis



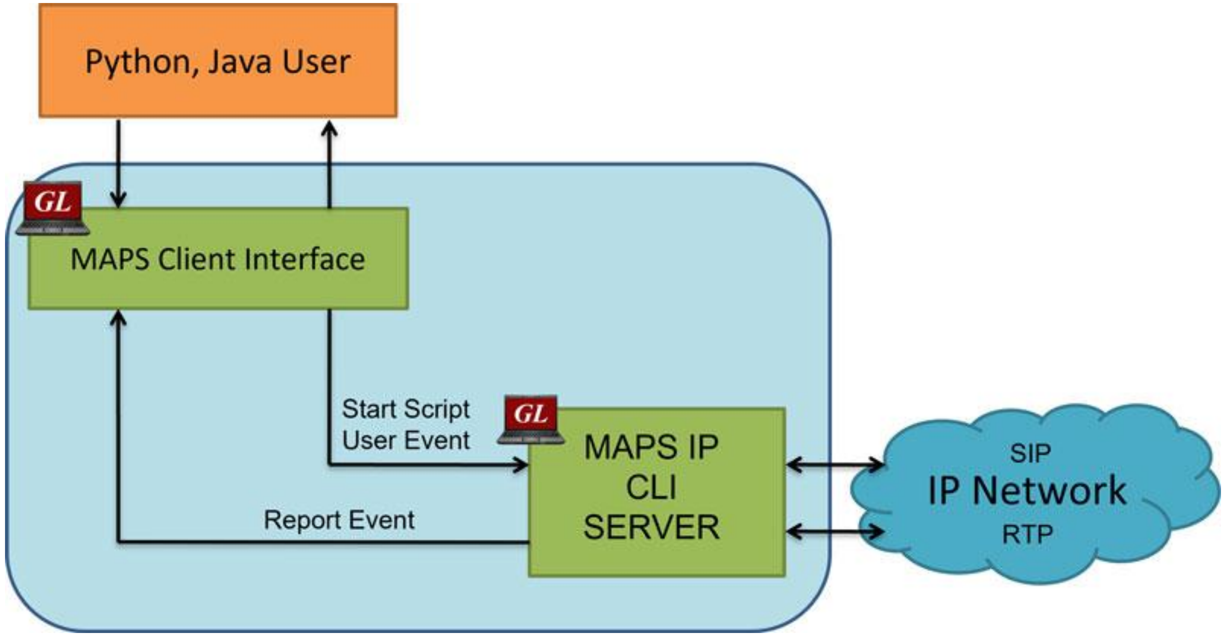
MAPS™ Command Line Interface



- MAPS™ can be configured as server-side application, to enable remote controlling through multiple command-line based clients. Supported clients include Python and Java
- The MAPS™ APIs allows for programmatic and automated control over all MAPS™ platforms. Each MAPS™ server can receive multiple client connections and offer independent execution to each client
- Likewise, a single client can connect to multiple MAPS™ servers, including servers running different protocols, permitting complex cross-protocol test cases

MAPS™ SIP CLI Test System

- As depicted, MAPS™ SIP CLI test system consists of the following -
  - Python, Java user communicating over TCP/IP
  - MAPS™ Client IFC, and MAPS™ SIP CLI Server



GL Communications

MAPS™ CLI Server and Python Client

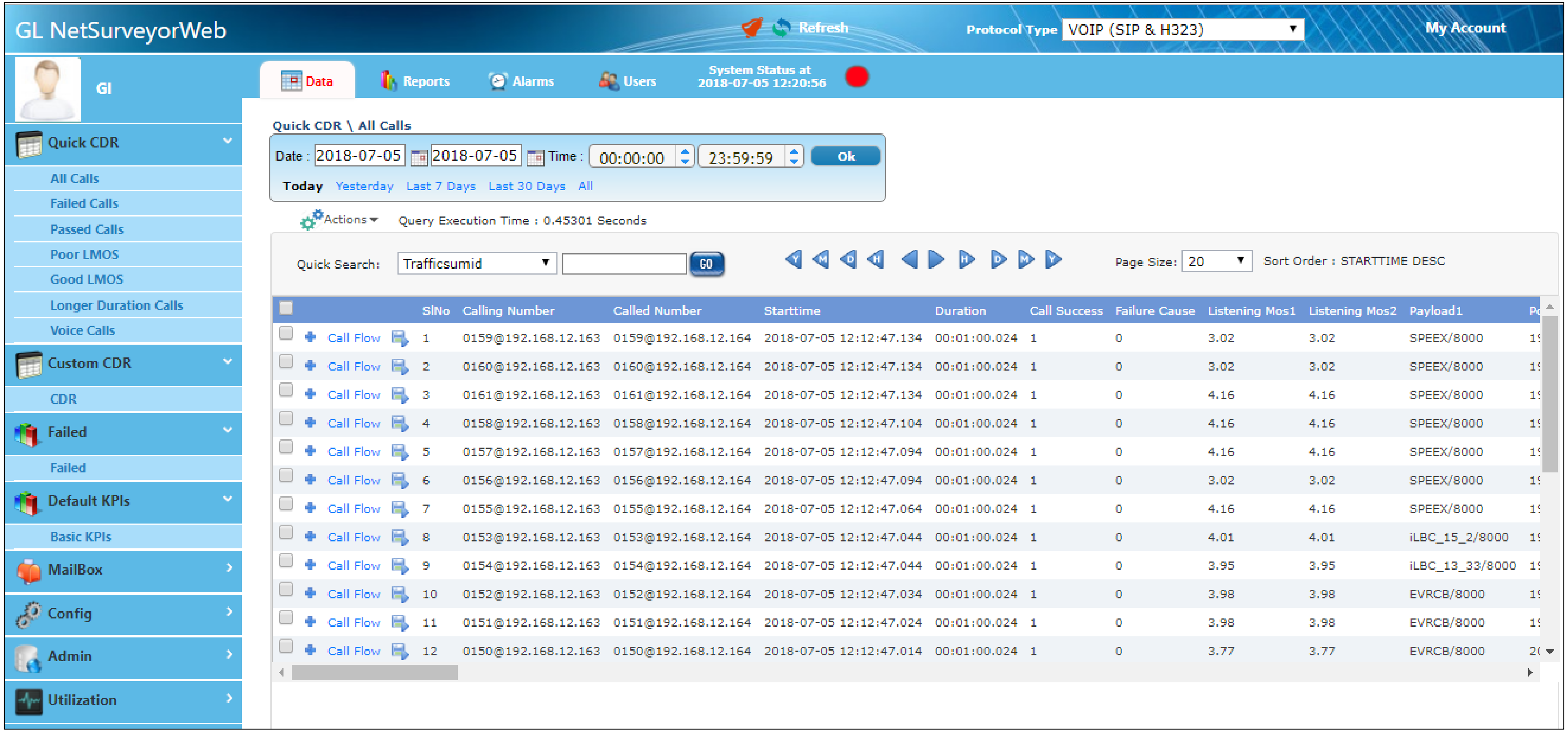
```
MapsCLI (SIP IETF )
File  Edit  View
View Latest Command

5 :: 2020-7-3 13:06:18.333000 : Start "TestBedDefault.xml" ;
5 :: 2020-7-3 13:06:18.442000 : LoadProfile "UserAgent_Profiles.xml"
5 :: 2020-7-3 13:06:18.770000 : Apply Global Configuration # "_EnableCLI"=1;
5 :: 2020-7-3 13:06:18.771000 : StartScript 1 "SipCallControl.gls" "Profile0001" 1 ;
5 :: 2020-7-3 13:06:18.880000 : UserEvent 1 "SetVariable"# "Contact"="1231230001@192.168.12.216";
5 :: 2020-7-3 13:06:18.991000 : UserEvent 1 "SetVariable"# "AddressOfRecord"="1231230001@192.168.12.216";
5 :: 2020-7-3 13:06:19.105000 : UserEvent 1 "SetVariable"# "RtpIpAddress"="192.168.12.216";
5 :: 2020-7-3 13:06:19.209000 : UserEvent 1 "SetVariable"# "To"="0001@192.168.12.209";
5 :: 2020-7-3 13:06:19.318000 : UserEvent 1 "SetVariable"# "Packetizationtime"="20";
5 :: 2020-7-3 13:06:19.429000 : UserEvent 1 "SetVariable"# "OvrCodecListSize"=3;
5 :: 2020-7-3 13:06:19.540000 : UserEvent 1 "SetVariable"# "OvrCodecList[0]"="G729";
5 :: 2020-7-3 13:06:19.646000 : UserEvent 1 "SetVariable"# "OvrPayloadList[0]"=18;
5 :: 2020-7-3 13:06:19.756000 : UserEvent 1 "SetVariable"# "OvrCodecList[1]"="PCMU";
5 :: 2020-7-3 13:06:19.864000 : UserEvent 1 "SetVariable"# "OvrPayloadList[1]"=0;
5 :: 2020-7-3 13:06:19.979000 : UserEvent 1 "SetVariable"# "OvrCodecList[2]"="telephone-event";
5 :: 2020-7-3 13:06:20.085000 : UserEvent 1 "SetVariable"# "OvrPayloadList[2]"=101;
5 :: 2020-7-3 13:06:20.192000 : UserEvent 1 "RTP_CreateSession";
5 :: 2020-7-3 13:06:24.349000 : UserEvent 1 "GetCallStatus";
5 :: 2020-7-3 13:06:24.460000 : UserEvent 1 "GetNegotiatedCodec";
5 :: 2020-7-3 13:06:24.569000 : UserEvent 1 "SendFile"# "TxFileName"="voicefiles\Send\G711\ULAW\Vijay.glw","TxFileDuration"=10;
5 :: 2020-7-3 13:06:34.635000 : UserEvent 1 "GetVoiceQualityStats";
5 :: 2020-7-3 13:06:34.739000 : UserEvent 1 "SIP_TerminateCall";
5 :: 2020-7-3 13:06:34.848000 : UserEvent 1 "GetMessageCount";
5 :: 2020-7-3 13:06:34.957000 : UserEvent 1 "GetMessageInfo"# "Index"=0;
5 :: 2020-7-3 13:06:35.067000 : UserEvent 1 "GetMessageInfo"# "Index"=1;
5 :: 2020-7-3 13:06:35.178000 : UserEvent 1 "GetMessageInfo"# "Index"=2;
5 :: 2020-7-3 13:06:35.290000 : UserEvent 1 "GetMessageInfo"# "Index"=3;
5 :: 2020-7-3 13:06:35.397000 : UserEvent 1 "GetMessageInfo"# "Index"=4;
5 :: 2020-7-3 13:06:35.510000 : UserEvent 1 "GetMessageInfo"# "Index"=5;
5 :: 2020-7-3 13:06:35.616000 : UserEvent 1 "GetMessageInfo"# "Index"=6;
5 :: 2020-7-3 13:06:35.724000 : StopScript 1;
ServerLog:errCode = 0,errString = connection has been gracefully closed for ClientId =5
```

```
Python 3.7.3 Shell
File  Edit  Shell  Debug  Options  Window  Help
Python 3.7.3 (v3.7.3:ef4ec6ed12, Mar 25 2019, 22:22:05) [MSC v.1916 64 bit (AMD64)] on win32
Type "help", "copyright", "credits" or "license()" for more information.
>>>
 RESTART: C:\Program Files\GL Communications Inc\MAPS-SIP\PythonClient\examples\SIP\SipBasicCall.py
SERVER INITIALIZED
CONNECTED
Negotiated Codec = PCMU
0
CMOS = 4.19531
LMOS = 4.19531
CR_FACTOR = 93
LR_FACTOR = 93
TX_PACKETS = 501
RX_PACKETS = 712
LOST_PACKETS = 0
DISCARDED_PACKETS = 0
OUT_OF_SEQ_PACKETS = 0
DUPLICATE_PACKETS = 0
AVG_JITTER = 0.125

12:24:01.120     ->      INVITE
INVITE sip:0001@192.168.12.209 SIP/2.0
Via: SIP/2.0/UDP 192.168.12.216:5060;branch=z9hG4bK-4-1348328288-22704-17372
Max-Forwards: 70
Allow: INVITE,BYE,CANCEL,ACK,INFO,OPTIONS,SUBSCRIBE,NOTIFY,REFER,REGISTER,UPDATE
From: 1231230001 <sip:1231230001@192.168.12.216>;tag=FromTag-1-1348328288-22701-17372
To: 0001 <sip:0001@192.168.12.209>
Call-ID: GL-MAPS-3-1348328288-22703-17372@192.168.12.216
CSeq: 1 INVITE
Contact: 1231230001 <sip:1231230001@192.168.12.216>
Content-Type: application/sdp
Content-Length: 269

v=0
o=1231230001 39377840 1 IN IP4 192.168.12.216
s=SIP Call
c=IN IP4 192.168.12.216
t=0 0
m=audio 1024 RTP/AVP 18 0 101
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
```

GL Communications

NetSurveyorWeb™

- Multiple PacketScan™ probes can be used for network monitoring, with call detail reports exported to a central database
- Results can be accessed remotely using NetSurveyorWeb™, a simple web browser based application



GL Communications

Thank you